



## **SIP-SDP Inter-IMS NNI Profile**

**Version 9.0**

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# 1 Introduction

## 1.1 Overview

This document describes a SIP/SDP profile for interconnection and roaming NNI between operators IMS networks for the purposes of exchanging traffic originating from and terminating to the respective operators' customers. This document profiles SIP/SDP for the GSMA defined IMS based services ((VoLTE Voice over LTE, Video Call, IMS data Channel, SMSoIP (SMS over IP), and RCS Rich Communications Services services) as described in the relevant GSMA PRDs cited in clause 2.

This profile is intended to be a generic NNI profile that may be applied to any such inter-operator interconnect, including IPX and direct bilateral interconnect on regional and international basis, thereby promoting commonality and facilitating interoperability. Where options are supported based on bilateral agreement, such agreement is between the respective Operators or between Operator and IPX Provider dependent on the type of interconnect.

It is also acknowledged that some organizations may still wish to define national specific interconnect profiles. Whilst it is hoped that such national specific variants can be avoided, this document may also be used as a basis for such national specific variants and thereby minimising duplication of effort.

## 1.2 Relationship to Existing Standards

This document is a profile of 3GPP Release-11 TS 29.165 [1] and provides clarifications and recommendations to that technical specification to facilitate inter-IMS (IP Multimedia System) NNI (Network-Network Interface) inter-operability. The requirements are derived from GSMA PRDs for the provision of IMS-based voice, video and RCS services.

The VoLTE, Video Call, IMS Data Channel, SMSoIP, VoWiFi, and RCS services are described in the following GSMA PRDs:

- IR.92 – IMS Profile for Voice and SMS [2]
- IR.94 – IMS Profile for Conversational Video Service [3]
- RCC.07 – RCS Advanced Communications Services & Client Specification [44]
- IR.51 – IMS Profile for Voice, Video, and SMS over untrusted WiFi [91]
- NG.106 – IMS Profile for Voice, Video, and SMS over trusted WiFi [8]
- NG.134 – IMS Profile for Data Channel [130]

The NNI aspects are described in the following GSMA PRDs:

1. IR.65 – IMS Roaming NNI and Interworking Guidelines [4]
2. IR.88 – LTE Roaming NNI Guidelines [6]
3. IR.90 – RCS Interworking Guidelines [5]

### 1.3 Scope

This document specifies a SIP/SDP profile across the inter-IMS NNI in support of VoLTE, Video Call, IMS Data Channel, SMSoIP, VoWiFi, and RCS services. The VoLTE and Video Call Services are based on 3GPP MMTel, specifically the sub-set of MMTel services as described in GSMA PRDs IR.92 [2] and IR.94 [3]. In the remainder of this document, the Voice and Video Call supplementary services shall be referred to as MMTel services.

The exact set of services to be supported is determined by mutual agreement between operators. Where a reduced set of services is agreed, an appropriate subset of this profile is applicable.

There are two aspects for the NNI profiled within this document, namely the Interconnect NNI and the Roaming NNI which are defined as below:

- Interconnect-NNI – This term applies when the NNI is used to exchange traffic between the serving network (home or visited network) of the originating device, and the home network of the called party.
- Roaming NNI – This term applies when the NNI is used to exchange traffic between the home and visited networks of a roaming device, i.e. when using Local Break-Out (LBO) where the P-CSCF is in the Visited Network. It is also inclusive of a scenario whereby the IMS core network is under different administrative control to the home Operator’s access network (i.e. hosted solution).

Note: The use of the specification in support of fixed line access is not precluded.

### 1.4 Definition of Acronyms and Terms

#### 1.4.1 Acronyms

Term	Description
3GPP	3rd Generation Partnership Project
AMR	Adaptive Multi-Rate
AMR-WB	AMR Wide Band
CONF	Conference (MMTel Service)
CPM	Converged IP Messaging
DTLS	Datagram Transport Layer Security
DTMF	Dual Tone Multi Frequency
ECT	Explicit Communication Transfer (MMTel Service)
EVRC	Enhanced Variable Rate Codec
EVS	Enhanced Voice Services
GSMA	GSM Association
IBCF	Interconnection Border Control Function
IETF	Internet Engineering Task Force
II-NNI	Inter-IMS Network to Network Interface
IMDN	Instant Message Delivery Notification
IMS	IP Multimedia Subsystem

<b>Term</b>	<b>Description</b>
IP	Internet Protocol
IPX	IP Exchange
ISDN	Integrated Services Digital Network
ITU	International Telecommunications Union
ITU-T	Telecoms Standardization Sector of ITU
LTE	Long Term Evolution
MaaP	Messaging as a Platform
MCID	Malicious Call Identification (MMTel Service)
MIME	Multipurpose Internet Mail Extensions
MMTel	Multimedia Telephony
MSRP	Message Sending Relay Protocol
NNI	Network-Network Interface
OMA	Open Mobile Alliance
PRD	Permanent Reference Document
RAVEL	Roaming Architecture for Voice over IMS with Local Breakout
RCS	Rich Communications Services
RFC	Request For Comments
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SMS	Short Messaging Service
SIMPLE	SIP for Instant Messaging and Presence Leveraging Extensions
SRVCC	Single Radio Voice Call Continuity
TCP	Transmission Control Protocol
TrGW	Transition Gateway
TS	Technical Specification
UCE	UE Capability Exchange
UDP	User Datagram Protocol
UFA	UCE Forking & Aggregation
UP	Universal Profile
VBD	Voice Band Data
VoLTE	Voice over LTE
VoWiFi	Voice over Wi-Fi
XDM	XML Document Management
XML	Extended Mark-up Language
WiFi	Wireless Fidelity

## 1.4.2 Terms

Term	Description
Confirming SDP Offer	The second SDP offer of the two offer/answer exchanges that is possibly required if preconditions are used. This second SDP offer is started by the originating or terminating UE, e.g., with the SDP of the SIP Update request, is used for precondition status update (see GSMA PRD IR.92 [2] and confirms the selected RTP payload type.

## 1.5 Document Cross-References

Ref	Doc Number	Title
[1]	3GPP TS 29.165 R11	Inter-IMS Network-Network Interface (NNI)
[2]	GSMA PRD IR.92	IMS Profile for Voice and SMS
[3]	GSMA PRD IR.94	IMS Profile for Conversational Video Service
[4]	GSMA PRD IR.65	IMS Roaming NNI and Interworking Guidelines
[5]	GSMA PRD IR.90	RCS Interworking Guidelines
[6]	GSMA PRD IR.88	LTE Roaming NNI Guidelines
[7]	GSMA PRD IR.74	Video Share Interoperability Specification
[8]	GSMA PRD NG.106	IMS Profile for Voice, Video and SMS over trusted Wi-Fi
[9]	3GPP TS 26.114	IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction
[10]	GSMA PRD NG.102	IMS Profile for Converged IP Communications
[11]	IETF RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP)
[12]	IETF RFC 3261	Session Initiation Protocol (SIP)
[13]	IETF RFC 4566	Session Description Protocol (SDP)
[14]	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
[15]	IETF RFC 3312	Integration of Resource Management and Session Initiation Protocol (SIP)
[16]	IETF RFC 4028	Session Timers in the Session Initiation Protocol (SIP)
[17]	3GPP TS 29.238	Interconnection Border Control Function (IBCF) – Transition Gateway (TrGW) interface, Ix Interface; Stage 3
[18]	IETF RFC 3556	SDP Bandwidth Modifiers for RTCP bandwidth
[19]	IETF RFC 3891	The Session Initiation Protocol (SIP) “Replaces” Header

Ref	Doc Number	Title
[20]	IETF RFC 3327	Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts
[21]	IETF RFC 4488	Suppression of Session Initiation Protocol (SIP) REFER Method Implicit Subscription
[22]	IETF RFC 4733	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
[23]	GSMA PRD IR.67	DNS Guidelines for Service Providers and GRX/IPX Providers
[24]	ITU-T Rec. T.140 (1998)	Text Conversation Protocol for multimedia application, with amendment 1 (2000)
[25]	GSMA PRD NG.118	NNI Parameter Forms
[26]	3GPP TS 33.210	3G Security; Network Domain Security (NDS); IP Network Layer Security
[27]	IETF RFC 4303	IP Encapsulating Security Payload (ESP)
[28]	IETF RFC 5368	Referring to Multiple Resources in the Session Initiation Protocol (SIP)
[29]	ITU-T Rec. H.263 (2005)	Video Coding for low bit rate communication
[30]	IETF RFC 3984	RTP Payload format for ITU-T Rec. H.264 Video
[31]	Void	
[32]	IETF RFC 3551	RTP Profile for Audio Video Conferences with Minimal Control
[33]	IETF RFC 4585	Extended RTP Profile for Real Time Control Protocol (RTCP)-Based Feedback (RTP/AVPF)
[34]	OMA CPM	OMA CPM Conversation Functions (OMA-CPM-TS_Conv_Func-V1_0-20120612-A)
[35]	IETF RFC 5939	Session Description Protocol (SDP) Capability Negotiation
[36]	Void	
[37]	3GPP TS 29.079	Optimal Media Routing within the IP Multimedia System (IMS); Stage 3
[38]	IETF RFC 5621	Message Body Handling in the Session Initiation Protocol (SIP)
[39]	IETF RFC 5547	A Session Description Protocol (SDP) Offer/Answer Mechanism to enable File Transfer
[40]	IETF RFC 4483	A Mechanism for Content Indirection in Session Initiation Protocol (SIP) Messages
[41]	IETF RFC 5438	Instant Message Disposition Notification
[42]	IETF RFC 3842	A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)



Ref	Doc Number	Title
[43]	IETF RFC 4575	A Session Initiation Protocol (SIP) Event Package for Conference State
[44]	GSMA PRD RCC.07	RCS 7.0 Advanced Communications Services & Client Specification
[45]	3GPP TS 29.163	Interworking between IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched networks
[46]	IETF RFC 3966	The Tel URI for telephone numbers
[47]	3GPP TS 24.341	Support of SMS over IP networks; Stage 3
[48]	IETF RFC 4244	An extension to the Session Initiation Protocol (SIP) for Request History Information
[49]	IETF RFC 4916	Connected Identity in the Session Initiation Protocol (SIP)
[50]	IETF RFC 3680	A Session Initiation Protocol (SIP) Event Package for registration
[51]	IETF RFC 3515	The Session Initiation Protocol (SIP) Refer method
[52]	IETF RFC 3840	Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)
[53]	IETF RFC 3856	A Presence Event Package for the Session Initiation Protocol (SIP)
[54]	IETF RFC 3857	A Watcher-Information Event Template Package for the Session Initiation Protocol (SIP)
[55]	IETF RFC 3858	An Extensible Mark-Up Language (XML) based format for Watcher Information
[56]	IETF RFC 3863	Presence Information Data Format (PIDF)
[57]	IETF RFC 4479	A data model for Presence
[58]	IETF RFC 4975	The Message Session Relay Protocol (MSRP)
[59]	IETF RFC 5364	Extensible Markup Language (XML) Format Extension for Representing Copy Control Attributes in Resource Lists
[60]	IETF RFC 5366	Conference Establishment Using Request-Contained Lists in the Session Initiation Protocol (SIP)
[61]	IETF RFC 5368	Referring to Multiple Resources in the Session Initiation Protocol (SIP)
[62]	3GPP TS 29.658	SIP Transfer of IP Multimedia Tariff Information; Protocol Specification.
[63]	3GPP TS 24.337	IP Multimedia (IM) Core Network (CN) Subsystem; IP Multimedia Subsystem (IMS) Inter UE Transfer; Stage 3
[64]	3GPP TS 24.237	IP Multimedia (IM) Core Network (CN) Subsystem; IP Multimedia Subsystem (IMS) Service Continuity; Stage 3
[65]	IETF RFC 5627	Obtaining and using Globally Routable User Agent URIs (GRUUs) in the Session Initiation Protocol (SIP)

Ref	Doc Number	Title
[66]	IETF RFC 4412	Communications Resource Priority in the Session Initiation Protocol (SIP).
[67]	3GPP TS 24.628	Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[68]	3GPP TS 24.229	IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3
[69]	3GPP TS 29.162	Interworking between the IMS CN Subsystem and IP Networks
[70]	3GPP TS 26.071	Mandatory speech CODEC speech processing functions; AMR speech Codec; General description
[71]	3GPP TS 26.090	Mandatory Speech Codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Transcoding functions
[72]	3GPP TS 26.073	ANSI C code for the Adaptive Multi Rate (AMR) speech codec
[73]	3GPP TS 26.104	ANSI-C code for the floating-point Adaptive Multi-Rate (AMR) speech codec
[74]	Void	
[75]	3GPP TS 26.171	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; General description
[76]	3GPP TS 26.173	ANSI-C code for the Adaptive Multi-Rate – Wideband (AMR-WB) speech codec
[77]	3GPP TS 26.190	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; Transcoding functions
[78]	3GPP TS 26.193	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; Source controlled rate operation
[79]	3GPP TS 26.204	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; ANSI-C code
[80]	3GPP TS 23.228	IP Multimedia Subsystem (IMS); Stage 2
[81]	IETF RFC 7329	A Session Identifier for the Session Initiation Protocol (SIP)
[82]	IETF RFC 6086	Session Initiation Protocol (SIP) INFO Method and Package Framework
[83]	3GPP TS 26.441	Codec for Enhanced Voice Services (EVS); General overview
[84]	3GPP TS 26.442	Codec for Enhanced Voice Services (EVS); ANSI C code (fixed-point)

Ref	Doc Number	Title
[85]	3GPP TS 26.443	Codec for Enhanced Voice Services (EVS); ANSI C code (floating-point)
[86]	3GPP TS 26.445	Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description
[87]	3GPP TS 26.447	Codec for Enhanced Voice Services (EVS); Error Concealment of Lost Packets
[88]	3GPP TS 26.449	Codec for Enhanced Voice Services (EVS); Comfort Noise Generation (CNG) Aspects
[89]	3GPP TS 26.450	Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)
[90]	3GPP TS 26.451	Codec for Enhanced Voice Services (EVS); Voice Activity Detection (VAD)
[91]	GSMA PRD IR.51	IMS Profile for Voice, Video and SMS over untrusted Wi-Fi
[92]	3GPP TS 23.334	IP Multimedia Subsystem (IMS) Application Level Gateway (IMS-ALG) - IMS Access Gateway (IMS-AGW) interface: Procedures descriptions
[93]	IETF RFC 4687	RTP Payload Format and File Storage Format
[94]	IETF RFC 6184	RTP Payload Format for H.264 Video
[95]	IETF RFC 3389	RTP Payload for Comfort Noise
[96]	GSMA PRD RCC.71	RCS Universal Profile Service Definition Document
[97]	GSMA PRD NG.105	ENUM Guidelines for Service Providers and IPX Providers
[98]	Void	
[99]	IETF RFC 3841	Caller Preferences for the Session Initiation Protocol (SIP)
[100]	IETF RFC 5104	Codec Control Messages in the RTP Audio-Visual Profile with Feedback (AVPF)
[101]	IETF RFC 5285	A General Mechanism for RTP Header Extensions
[102]	IETF RFC 6665	SIP-Specific Event Notification
[103]	IETF RFC 5373	Requesting Answering Modes for the Session Initiation Protocol (SIP)
[104]	IETF RFC 3310	Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)

Ref	Doc Number	Title
[105]	IETF RFC 6446	Session Initiation Protocol (SIP) Event Notification Extension for Notification Rate Control
[106]	IETF RFC 6809	Mechanism to Indicate Support of Features and Capabilities in the Session Initiation Protocol (SIP)
[107]	IETF RFC 5626	Managing Client-Initiated Connections in the Session Initiation Protocol (SIP)
[108]	IETF RFC 6442	Location Conveyance for the Session Initiation Protocol
[109]	IETF RFC 3911	The Session Initiation Protocol (SIP) "Join" Header
[110]	IETF RFC 5393	Addressing an Amplification Vulnerability in Session Initiation Protocol (SIP) Forking Proxies
[111]	IETF RFC 7315	Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3GPP
[112]	IETF RFC 4964	The P-Answer-State Header Extension to the Session Initiation Protocol for the Open Mobile Alliance Push to Talk over Cellular
[113]	IETF RFC 3325	Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
[114]	IETF RFC 6050	A Session Initiation Protocol (SIP) Extension for the Identification of Services
[115]	IETF RFC 5009	Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media
[116]	IETF RFC 7316	The Session Initiation Protocol (SIP) P-Private-Network-Indication Private Header (P-Header)
[117]	IETF RFC 5002	The Session Initiation Protocol (SIP) P-Profile-Key Private Header (P-Header)
[118]	IETF RFC 5318	The Session Initiation Protocol (SIP) P-Refused-URI-List Private-Header (P-Header)
[119]	IETF RFC 5502	The SIP P-Served-User Private-Header (P-Header) for the 3GPP IP Multimedia (IM) Core Network (CN) Subsystem
[120]	IETF RFC 5360	A Framework for Consent-Based Communications in the Session Initiation Protocol (SIP)
[121]	IETF RFC 6794	A Framework for Session Initiation Protocol (SIP) Session Policies

Ref	Doc Number	Title
[122]	IETF RFC 3323	A Privacy Mechanism for the Session Initiation Protocol (SIP)
[123]	IETF RFC 3326	The Reason Header Field for the Session Initiation Protocol (SIP)
[124]	IETF RFC 3892	The Session Initiation Protocol (SIP) Referred-By Mechanism
[125]	IETF RFC 3329	Security Mechanism Agreement for the Session Initiation Protocol (SIP)
[126]	IETF RFC 3608	Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration
[127]	IETF RFC 3903	Session Initiation Protocol (SIP) Extension for Event State Publication
[128]	IETF RFC 5839	An Extension to Session Initiation Protocol (SIP) Events for Conditional Event Notification
[129]	IETF RFC 4538	Request Authorization through Dialog Identification in the Session Initiation Protocol (SIP)
[130]	GSMA PRD NG.134	IMS Data Channel
[131]	IETF RFC 8841	Session Description Protocol (SDP) Offer/Answer Procedures for Stream Control Transmission Protocol (SCTP) over Datagram Transport Layer Security (DTLS) Transport
[132]	IETF RFC 8842	Session Description Protocol (SDP) Offer/Answer Considerations for Datagram Transport Layer Security (DTLS) and Transport Layer Security (TLS)
[133]	IETF RFC 8864	Negotiation Data Channels Using the Session Description Protocol (SDP)
[134]	IETF RFC 4572	Connection-Oriented Media Transport over the Transport Layer Security (TLS) Protocol in the Session Description Protocol (SDP)
[135]	IETF RFC 4145	TCP-Based Media Transport in the Session Description Protocol

Note: Unless otherwise stated, the latest version of the referenced specifications applies.

## 2 Applicable Services

As stated previously, this profile supports VoLTE, Video Call, IMS Data Channel, SMSoIP, VoWiFi, MMTEL Call Composer and RCS services. The RCS Services, as defined in GSMA PRD RCC.71 [96] include all of the following:

1. Capability Exchange based on SIP OPTIONS (Note 1)
2. Capability Exchange via Presence (Note 1)
3. 1:1 Chat/Group Chat
4. Standalone messaging
5. File Transfer based on HTTP (Hypertext Transfer Protocol)
6. MSRP based Enriched Calling Services (Call Composer, Call Unanswered, Shared Map, Shared Sketch)
7. Audio Messaging
8. Geo-location Push
9. Chatbot Sessions for MaaP
10. Extension to Extension services

The relevant feature tags that are applicable to the above service set are as listed in Section 2.4.4 of GSMA PRD RCC.07 [44].

As described in GSMA PRD IR.65 [4] section 5.7, an Operator can deploy: -

- a converged IMS core network supporting all services,
- two separate IMS core networks with respective service sub-sets provided via the two IMS core networks. One IMS core network supports voice, video, Data Channel, MMTEL Call Composer and SMSoIP whilst the other IMS core network supports RCS services.

The inter-working between different IMS core network deployments is discussed in section 13.

The inter-working between different Capability Exchange methods is discussed in section 14.

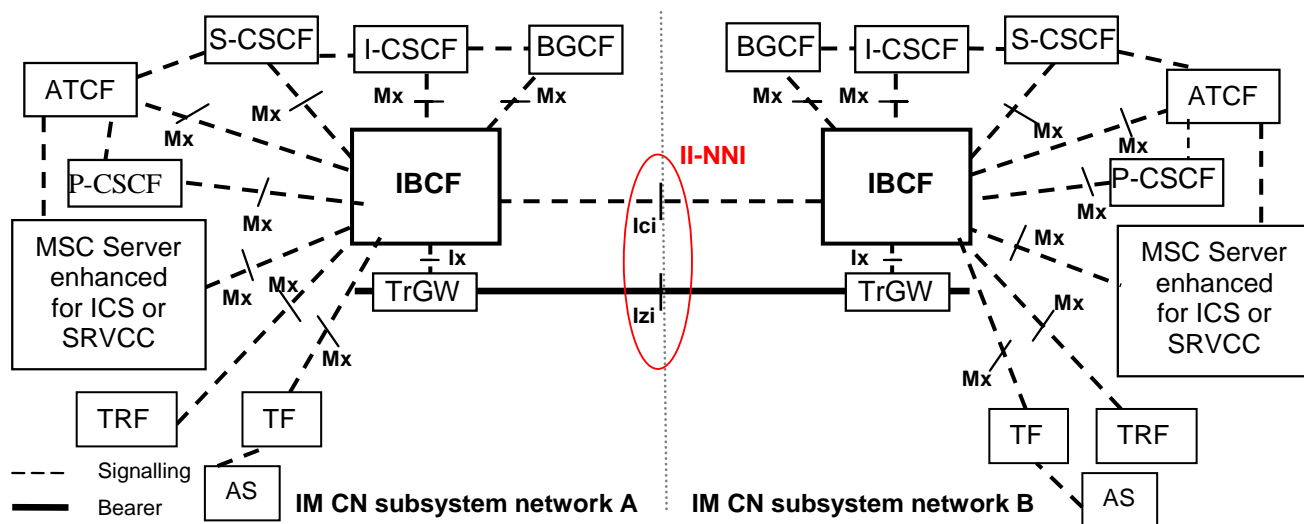
Note 1: If Capability Exchange via Presence is bilaterally agreed, then (Session Initiation Protocol) SIP OPTIONS need not to be supported for Capability Exchange, and vice versa.

Note 2: RCS also cites Voice call and Video call. However, these are based on GSMA PRDs IR.92 [2], IR.51 [91] and IR.94 [3] respectively.

## 3 IP Interconnection

IP interconnection is described in GSMA PRD IR.65 [4] and may be accomplished via IPX (IP Exchange) or else via direct point-point connectivity. Whichever option is chosen is determined by mutual agreement of the operators and has no impact on the SIP/SDP (Session Initiation Protocol /Session Description Protocol) profile across the NNI (Network-Network Interface). The IMS (IP Multimedia Subsystem) inter-NNI reference architecture (from 3GPP TS 29.165 [1]) is as shown in Figure 1 below. It is seen that the NNI interface consists of the Ici and Izi reference points between the peer IBCFs (Interconnection Border

Control Function) and TrGWs (Transition Gateway) in the control and media planes respectively.



**Figure 1 : Inter IMS NNI Reference Architecture**

As stated in 3GPP TS 29.165 [41], IMS roaming may be performed across the II-NNI (Inter-IMS Network to Network Interface) subject to agreements between the operators. Such use of the NNI (the roaming-NNI) is in scope for this profile.

#### 4 SIP Methods & Headers

Table 1 describes the SIP methods that are applicable in this profile. Each method is tagged as Mandatory / Optional where:

1. Mandatory means that the SIP method must be supported at each end,
2. Optional means that the SIP method must be supported dependent on a specific service or capability being applicable at the NNI. If no such service or capability is applicable, then the method is Not Applicable. Example services/capabilities are given. The list of example services/capabilities does not preclude others being applied via bilateral agreement.

Method	Status	Additional Information
INVITE	Mandatory	Includes both the initial INVITE and any subsequent re-INVITE
ACK	Mandatory	
BYE	Mandatory	
CANCEL	Mandatory	
OPTIONS	Mandatory	May be used for RCS Capability Exchange. May also be used as a heartbeat mechanism on the NNI.
INFO	Optional	May be used across the Roaming NNI for USSI (USSD over IMS).

Method	Status	Additional Information
		<p>May also be used for MMTel (Multimedia Telephony) features not included in GSMA PRD IR.92 [2] if bilaterally agreed (e.g. AOC, MCID (Malicious Call Identification (MMTel Service) etc. - see section 12 of 3GPP TS 29.165 [1])</p> <p>Older implementations (pre-IETF RFC 4028 [16]) may use INFO as a session heartbeat via bilateral agreement. In this case, a 200 OK response must be sent in reply to an INFO request that is syntactically correct and well structured, as defined in IETF RFC 6086 [82].</p>
MESSAGE	Mandatory	<p>Used for SMS over IP.</p> <p>Also used for RCS messaging (pager mode and IMDN, Instant Message Delivery Notification)</p>
NOTIFY	Mandatory	<p>Used for roaming NNI for “regevent” notification.</p> <p>Used at the interconnect NNI for MMTel services (e.g. CONF status), RCS Social Presence, Group Chat and Capability Exchange via Social Presence.</p>
PRACK	Mandatory	<p>PRACK shall be generated in response to any non-100 provisional responses to an INVITE message specifying the ‘100rel’ option tag in a Require header.</p> <p>PRACK may be generated in response to any non-100 provisional responses to an INVITE message specifying the ‘100rel’ option tag in a Supported header.</p> <p>Note that this may only be specified in the provisional response if the corresponding INVITE message indicated support of reliable provisional responses via inclusion of a ‘100rel’ option tag in either a Supported or Require header.</p>
PUBLISH	Mandatory	<p>Used for RCS Social Presence and applicable at the roaming NNI only.</p>
REFER	Mandatory	<p>Used for MMTel services (e.g. CONF, ECT) and RCS Group Chat.</p>
REGISTER	Mandatory	<p>Applicable for roaming NNI only.</p>
SUBSCRIBE	Mandatory	<p>Applicable for roaming NNI for “regevent”</p> <p>Also used at the interconnect NNI for RCS Social Presence, MMTel services (e.g. CONF status), RCS Group Chat and Capability Exchange via Social Presence.</p>
UPDATE	Mandatory	<p>Used for offer/answer exchange, session timer refresh etc.</p>

**Table 1: Applicable SIP Methods**



## **4.1 SIP Method Handling**

If a SIP method is received and recognized but not supported, it shall be rejected as defined in IETF RFC 3261 [12] with a SIP 405 "Method not allowed" response which shall include an ALLOW header field containing a list of supported methods.

If a SIP method is received and is not recognized (i.e. not implemented), it shall be rejected as defined in IETF RFC 3261 [12] by a 501 "Not Implemented" response.

Note: In order to prevent a given request being repeatedly re-sent, an IBCF may change a response code prior to forwarding the message across the NNI to a 403 "Forbidden" response.

## **4.2 SIP Status Code Handling**

SIP responses are handled according to IETF RFC 3261 [12].

As stated in IETF RFC 3261 [12], if a non-recognized final response is received in a SIP message then it shall be treated as being equivalent to the x00 response code of that class.

As stated in IETF RFC 3261 [12], if a non-recognized 18x provisional response (i.e. not referenced in the section 4.3.4.3, Table 3), is received in a SIP message, then it shall be treated as being equivalent to a 183 "Session Progress" response.

## **4.3 SIP Header Handling**

### **4.3.1 SIP Requests**

If a mandatory header is absent or malformed in the request, the request shall be rejected as defined in IETF RFC 3261 [12] with a SIP 400 "Bad Request" response.

If an unrecognized option tag is present in the Require header, the Request shall be rejected with a SIP 420 "Bad Extension" response. Other non-supported headers and parameters shall be ignored.

The headers or parameters that are not mentioned in the tables from Section 4.4 are considered as not applicable headers or parameters and shall be ignored as defined in IETF RFC 3261 [12].

### **4.3.2 SIP Responses**

If a header necessary for processing the response is absent or malformed in a final 2XX response to an INVITE request, the response shall be acknowledged by sending an ACK and then the dialog shall be terminated with a SIP BYE.

If a header necessary for processing the response is absent or malformed in a provisional response, the response shall be discarded.

If a header necessary for processing the response is absent or malformed in other final responses (i.e. except a 2XX response), the response shall be treated as the 500 "Server Internal Failure" response.

If a non-supported SIP header or parameter is received in a SIP response, it shall be ignored. Headers not listed in the subsequent tables in clause 4.4 and 4.5 are considered to be non-supported, unless there is a related bilateral agreement.

#### 4.4 SIP Header Support (Summary)

This clause summarises the SIP headers that are supported across the NNI across all SIP methods and responses. The supported headers are as documented in Table A.1 of 3GPP TS 29.165 [1] with additions as shown in Table 2 below. All the cited additional headers in Table 2 below are applicable only to RCS services using OMA CPM.

Header	Status	Additional Information
Contribution-ID	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Conversation-ID	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
InReplyTo- Contribution-ID	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Message- Expires	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Message-UID	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Session- Replaces	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Note: Status meaning is as defined in Table A.2 of 3GPP TS 29.165 [1]		

**Table 2: Supported SIP Headers (Overall)**

##### 4.4.1 Trust Relationships

Section 6.1.1.3.1 of 3GPP TS 29.165 [1] identifies the SIP headers that are subject to trust relationships.

The basic assumption for an MNO interconnecting to another MNO or IPX service provider using this service profile should be that there is a trust relationship. Nevertheless, some services are mainly applied within a single network, and SIP header related to such service should therefore not be passed over the II-NNI.

Table 3 below provides guidelines for trust relationships over the II-NNI to be used as complement to Table 6.2 of 3GPP TS 29.165 [1].

Note: The guidelines provided has taken an international II-NNI as the basis, when applied within a country, national regulatory requirements may also need to be considered.

Item	Header Field	Suggested Trust Relationship	Comment
	P-Asserted-Identity	Trusted	Trust is mandatory at the roaming NNI

Item	Header Field	Suggested Trust Relationship	Comment
	P-Access-Network-Info	Trusted	Should be trusted at the roaming NNI, even if “Not Trusted” at the non-roaming NNI
	Resource-Priority	Not Trusted	
	History-Info	Trusted	
	P-Asserted-Service	Trusted	Mandatory for terminating sessions delivered over Roaming NNI, when using LBO architecture [4] for all services that have an ISCI defined.
	P-Charging-Vector	Trusted	
	P-Charging-Function-Addresses	Not Trusted	
	P-Profile-Key	Not Trusted	Shall always be “Not Trusted” at a non-Roaming NNI
	P-Private-Network-Indication	Not Trusted	
	P-Served-User	Trusted Not Trusted	Shall always be “Trusted” at the Roaming-NNI Shall always be “Not Trusted” at a non-Roaming NNI
	Reason	Trusted	
	P-Early-Media	Trusted	
	Feature-Caps	Trusted	Trust is mandatory at the roaming NNI

**Table 3: Guidelines for trust relationship for SIP headers at the II-NNI**

#### 4.5 SIP Header Support (Per Method / Response)

This clause provides header details on a per SIP Method / Response basis. The header details per method/response are as tables B.3.1 through B.16.2 in 3GPP TS 29.165 [1] with any modifications/clarifications as described in this section.

##### 4.5.1 Additional Headers

There are a number of headers that are applicable only to RCS services using CPM that are defined in OMA CPM [34] and are not present in 3GPP TS 29.165 [1]. Table 4 below lists these headers and the methods/responses in which they may occur.

Header	Related Methods / Responses
Contribution-ID	INVITE request OR MESSAGE request Both with status do
Conversation-ID	INVITE request OR MESSAGE request – both with status do

Header	Related Methods / Responses
InReplyTo- Contribution-ID	INVITE request OR MESSAGE request – both with status do
Message-Expires	INVITE request – with status do
Message-UID	MESSAGE request OR 200 OK (MESSAGE) response OR BYE request – all with status do.
Session-Replaces	INVITE request – with status do
Note: Status meaning is as defined in Table B.2.1 of 3GPP TS 29.165 [1]	

**Table 4: Additional Headers**

Other headers that are not cited may be transited or removed at the NNI based on local operator policy and/or bi-lateral agreement.

#### 4.5.2 Header Manipulation

This section provides options/mandated actions on the manipulation of certain headers by the IBCF at the NNI. Note that the listed headers are not an exhaustive list and that header manipulation by the IBCF is dependent on operator policy.

Header	Header Modification / Manipulation
Call-ID	May be overwritten by the IBCF for topology hiding at the Interconnect-NNI but must not be overwritten at the Roaming NNI (see section 5.10.1 of 3GPP TS 24.229 [68]).
Content-Length	IBCF shall recalculate this number when message bodies are altered (e.g. SDP).
From	Can be anonymised by the IBCF based on the Privacy header.
Max-Forwards	May be overwritten by the IBCF.
Record-Route	This header may be overwritten for topology hiding.
Route	IBCF may include the identity of its peer in this header. For Roaming NNI, this header may also include the identity of HPLMN S-CSCF or VPLMN P-CSCF.
Via	May be overwritten by the IBCF for topology hiding.

**Table 5: IBCF Header Manipulation**

## 5 SIP Message Transport

Both UDP (User Datagram Protocol) and TCP (Transmission Control Protocol) transport are mandatory. If UDP is the transport of choice, then TCP should be used for large messages to avoid fragmentation as described in clause 18.1.1 of IETF RFC 3261 [12]. As stated in clause 18.1.1 of IETF RFC 3261 [12], it is recommended that an existing TCP connection be reused if a request is destined to an IP address, port, and transport to which an existing connection is already open.

SCTP (Stream Control Transmission Protocol) is optional and may be used if bilaterally agreed between the operators. As stated in 3GPP TS 29.165 [1], this option is favourable if the operators would like to improve reliability over the ICI interface.

Dependent on the nature of the connection, security may be applied at the NNI based on bilateral agreement. As stated in 3GPP TS 29.165 [1], the security mechanisms are as defined in 3GPP TS 33.210 [26]. It is recommended to use Network Domain Security (NDS) for IMS Protocols as defined in Annex C of 3GPP TS 33.210 [26] which mandates the use of IPSEC ESP [27].

## 6 SIP Signalling Mode

En-bloc signalling shall apply to this profile. The format of the address is described in clause 7.

## 7 Numbering & Addressing

The routing of SIP (Session Initiated Protocol) signaling over the IMS NNI requires the use of SIP URIs or tel URIs routable at the NNI per clause 6.4 of GSMA PRD IR.65 [4]. Five scenarios are possible for outgoing SIP sessions:

1. An identifier used for routing at the NNI may be formatted as a SIP URI whose user part is based on an E.164 format Public User Identity, and whose domain part is routable at the NNI. The “user=phone” parameter must be appended to such a URI.
2. An identifier used for routing at the NNI may be formatted as a SIP URI whose user part is alphanumeric and whose domain part is routable at the NNI.
3. An identifier used for routing at the NNI may by bilateral agreement be formatted as a SIP URI whose user part is based on a local telephone number (as defined in IETF RFC 3966 [46]), and whose domain part is routable at the NNI. In this case, the local number must be qualified via the phone-context parameter as defined in IETF RFC 3966 [46] which may be specified in terms of its global number (e.g. “+44”) or via a domain name. The format of the permitted phone-context is also subject to a bi-lateral agreement. The “user=phone” URI parameter must also be appended in this case.
4. An identifier used for routing at the NNI may be formatted as a tel URI containing E.164 format Public User Identity.
5. An identifier used for routing at the NNI may, if agreed bilaterally, be defined as a tel URI containing a local telephone number (as defined in IETF RFC 3966 [46]). In this case, the local number must be qualified via the phone-context parameter as defined in IETF RFC 3966 [46], which may be specified in terms of its global number (e.g., “+44”) or via a domain name. The format of the permitted phone-context is also subject to bilateral agreement.

The resolution of an E.164 format Public User Identity to one or more SIP URIs is described in section 4.2.3 of GSMA PRD NG.105 [97]. It is further recommended that Number Portability is handled as described in section 4.1.5 of GSMA PRD NG.105 [97].

In all cases, if a SIP URI is entered by the user, its domain part may have to be converted by the originating network in order to be routable at the NNI.

The originating Service Provider discovers the SIP point of contact (e.g. IMS IBCF) specified by the terminating Service Provider as described in clause 4.5.2 of GSMA PRD IR.67 [23].

As specified in IETF RFC 3261 [12], the application layer address to which a SIP message is delivered is identified in the Request-URI. To reach that address the message may traverse a sequence of SIP-aware network elements belonging to one or more networks.

To constrain its path, a “stack” of URIs may be encoded in one or more Route headers and appended to the SIP message. At each network element that performs SIP routing, a SIP-aware network element toward which the message is to be forwarded is identified by the URI at the top of that stack. A network element that owns the resource identified by the topmost URI removes that URI from the stack; and removes the associated Route header if it contains no additional URIs. If a message contains no Route headers, it is forwarded based on the URI in the Request line (i.e., the Request-URI).

## 8 SIP Message Bodies

3GPP TS 29.165 [1] states that the MIME (Multipurpose Internet Mail Extensions) type "application/sdp" and multipart message bodies (multipart/mixed, multipart/related and multipart/alternative) shall be supported according to IETF RFC 5621 [38] over the NNI.

3GPP TS 29.165 [1] also lists a number of other MIME types may be supported over the NNI based on agreement between operators. A number of these additional MIME types are related to MMTel or RCS services.

Table 6 below lists the MIME types that are recommended to be supported in this profile, based on the related services in scope across the NNI:

MIME Type	Additional Info
application/SDP	Mandatory. Used to carry SDP bodies to describe MMTel audio/video sessions and RCS TCP/MSRP (Message Sending Relay Protocol) sessions. The IBCF (in conjunction with information received from the TrGW) manipulates SDP message bodies.
multipart/mixed	Mandatory to align with 3GPP TS 29.165 [1]. Used in RCS messaging where multiple message bodies are included to send an initial message as well as negotiate a TCP/MSRP session. The IBCF manipulates the SDP to reflect the TCP/MSRP session traversing the TrGW.
multipart/related	Mandatory to align with 3GPP TS 29.165 [1]. Used in RCS FT to enable multiple message bodies to be included to both negotiate a TCP/MSRP session and include a thumbnail file preview (see IETF RFC 5547 [39]).
multipart/alternative	Despite being mandatory in 3GPP TS 29.165 [1], not specifically used for MMTel or RCS at the current time. Need not be manipulated by the IBCF. May be transited or removed by the IBCF based on operator preference.
message/external-body	Used in RCS messaging/FT to pass a reference to stored content, identified via a URI (see IETF RFC 4483 [40]).

MIME Type	Additional Info
	Conditionally supported in this profile of RCS messaging/FT is in scope across the NNI. Conveyed unchanged by the IBCF.
message/cpim	Used in RCS standalone (pager mode) messaging. Conditionally supported in this profile if RCS messaging is in scope across the NNI. Conveyed unchanged by the IBCF.
message/imdn+xml	Used in RCS messaging to inform the sender of message delivery/read (see IETF RFC 5438 [41]). Conditionally supported in this profile if RCS messaging is in scope across the NNI. Conveyed unchanged by the IBCF.
application/vnd.etsi.mcid+xml	Used in the MMTel MCID service (see 3GPP TS 24.616). This service is not mandated in GSMA PRD IR.92 [2] and this message body is thus optional in this profile and may be supported if bilaterally agreed.
application/vnd.3gpp.cw+xml	Used in n/w based Communication Waiting, which is not required in GSMA PRD IR.92 [2] and this message body is thus optional in this profile and may be supported if bilaterally agreed.
application/vnd.3gpp.comm-div-info+xml	Used in diversion notification, which is not required in GSMA PRD IR.92 [2] and this message body thus optional in this profile and may be supported if bilaterally agreed.
application/vnd.etsi.aoc+xml	Used for MMTel Advice of Charge Service, which is not required in GSMA PRD IR.92 [2] and this message body is thus optional in this profile and may be supported if bilaterally agreed.
application/vnd.etsi.cug+xml	Used for MMTel CUG Service, which is not required in GSMA PRD IR.92 [2] and this message body is thus optional in this profile and may be supported if bilaterally agreed.
application/vnd.etsi.sci+xml	Used for the transfer of real time charge information between the Charge Determination Point and Charge Recording Point (see 3GPP TS 29.658 [62]). Optional and may be supported if bilaterally agreed.
application/vnd.etsi.pstn+xml	Used to convey ISDN information (see 3GPP TS 29.163 [45]). This is conditionally supported where IMS is used as a transit network to connect CS-networks.
message/sipfrag	Used to convey SIP session progress. This is conditionally supported if MMTel Ad-Hoc Conference / RCS Group Chat service is used.
application/x-session-info	Used to convey additional digits in a SIP INFO for overlap sending. Not applicable to this profile.
application/pidf+xml, application/pidf-diff+xml,	Conditionally supported in RCS Social Presence is applicable at the NNI.

MIME Type	Additional Info
application/watcherinfo+xml, application/xcap-diff+xml, application/vnd.oma.suppnot+xml, application/simple-filter+xml	
application/resource-lists+xml, application/rlmi+xml	Used to convey a list of target users for MMTel Ad-Hoc Conference & RCS Group Chat.
application/load-control+xml	Used to exchange overload control information. The related internet draft is not yet agreed. Therefore, this is optional for this profile and may be supported if bilaterally agreed.
application/im-iscomposing+xml	Used to convey SIMPLE IM. Conditionally supported for RCS messaging services for interworking between SIMPLE IM and CPIM.
application/simple-message-summary+xml	Conditionally supported at the roaming NNI if the MMTel Message Waiting service is used. This service is included in GSMA PRD IR.92 [2].
application/vnd.3gpp.sms	Conditionally supported at the NNI if the SMS over IP service is in scope – see 3GPP TS 24.341 [47].
application/vnd.3gpp.ussd	Used for MMI at the roaming NNI. Optional to this profile and may be supported if bilaterally agreed.
application/vnd.3gpp.iut+xml application/vnd.3gpp.replication+xml	Used for inter-UE transfer. Optional in this profile and may be supported if bilaterally agreed. See 3GPP TS 24.337 [63].
application/vnd.3gpp.access-transfer-events+xml, application/vnd.3gpp.mid-call+xml, application/vnd.3gpp.srvcc-ext+xml, application/vnd.3gpp.srvcc-info+xml, application/vnd.3gpp.state-and-event-info+xml	Applicable to the roaming NNI and used for SRVCC (Single Radio Voice Call Continuity). Optional in this profile and may be supported if bilaterally agreed. See 3GPP TS 24.237 [64].
application/3gpp-ims+xml	Generic 3GPP XML body. This is optional in this profile and may be passed unaltered by the IBCF subject to bilateral agreement at the NNI.
application/reginfo+xml	Conditionally supported for the roaming NNI.
application/conference-info+xml	Conditionally supported if conference services are supported across the NNI (e.g. MMTel Ad-Hoc Conference, RCS Group Chat).

**Table 6: SIP Message Bodies**

The IBCF is primarily a transit point and may manipulate the application/SDP message body (e.g., due to its interaction with the TrGW to enable media flows to traverse the TrGW, and optionally to facilitate the media transcoding as described in 3GPP TS 23.228, Annex I, Section I.3.3 [80]).



If media transcoding is provided, then it shall be done as described in Section 10.3 of this GSMA PRD.

All other permitted message bodies are transited unchanged. The permitted MIME type of message bodies are selected based on local operator policy and/or bi-lateral agreement at the NNI dependent on the services supported at the NNI.

The IBCF may limit the size of SIP message bodies and take remedial action should that size be exceeded. The limit is agreed on a bilateral basis. The remedial action as specified in clause 5.10.6.3 of 3GPP TS 24.229 [68].

Other (unrecognised) message bodies may be removed or transited based on local operator policy and/or bi-lateral agreement at the NNI. The default action in this profile would be to remove such a message body.

## 9 SIP Options Tags

SIP Option tags are not removed when transited across the NNI.

Table 6.1.3.1 in 3GPP TS 29.165 [1] provides a list of capabilities supported across the NNI, a number of which are related to the use of SIP Option tags. All of the cited option tags are included in Table 6.1.3.1 of 3GPP TS 29.165 [1].

Table 7 below provides a list of the SIP Option Tags applicable to the scope/services of this profile based on GSMA PRD IR.65 [4] and GSMA PRD IR.90 [5] and carried in the SIP Supported or Require headers.

This list below in Table 7 does not preclude other tags being transmitted across the NNI nor a given tag being used in relation to other services.

Tag	Additional Information
timer	Enables SIP session keep-alive – see IETF RFC 4028 [16].
100rel	Enables reliable provisional responses – see IETF RFC 3262 [14]. An example would be conveying SDP in a 18X response.
precondition	Enables negotiation of resource reservation for segmented QOS – see IETF RFC 3312 [15]
path	Used for the roaming NNI scenario as part of IMS registration – see IETF RFC 3327 [20].
replaces	Applicable to MMTel services CONF and ECT – see IETF RFC 3891 [19].
histinfo	Used in MMTel Call Forwarding – see IETF RFC 4244 [48].
multiple-refer	Used for the roaming NNI when referring to multiple parties to be added to an existing conference – see IETF RFC 5368 [28]

Tag	Additional Information
norefersub	Used for the roaming NNI and enables implicit subscription to be notified of the progress of the associated REFER – see IETF RFC 4488 [21]
from-change	Related to the TIP service – see IETF RFC 4916 [49].
gruu	Used for the roaming NNI as party of IMS registration – see IETF RFC 5627 [65].
recipient-list-invite	Used for the roaming NNI when creating a conference via a list of URIs – see IETF RFC 5366 [60].
resource-priority	Used to denote priority for a SIP session – see IETF RFC 4412 [66].

**Table 7: SIP Option Tags**

## 10 Media Control

Media control shall follow the SIP SDP Offer/Answer model as documented in IETF RFC 3264 [11] to control the underlying user plane during a SIP session.

MMTel services shall negotiate voice, video or voice/video/data channel media flows. A single SIP session shall be able to negotiate a voice media flow, a video media flow or both a voice and video media flow. Video may be added to voice during a SIP session. All data channel media flow shall be negotiated and associated with the voice or voice/video media flows.

If RCS services are used (with the exception of pager-mode messaging, which has no user plane), TCP/MSRP sessions shall be negotiated to support the required media plane flows.

### 10.1 SIP SDP Offer / Answer

SIP SDP information shall be supported in the body of INVITE, ACK, PRACK, UPDATE, 200 OK (INVITE, PRACK, UPDATE) and 18x (INVITE) messages.

In line with GSMA PRD IR.92 [2], SIP preconditions must be supported by the UE but may be disabled by the network operator. It is out of scope of this profile whether the IBCF or some other network element is responsible for removing the related SDP media attributes to disable preconditions.

Irrespective of whether the network operator has disabled preconditions, the IBCF must support receiving SDP both with and without media attributes relating to preconditions.

If SIP preconditions are permitted, then the IBCF shall transit the related SDP media attributes to enable preconditions to be negotiated end-to-end, and thus two SDP offer/answer exchanges may be required to establish the bearer (e.g., INVITE/18x plus UPDATE/200 OK(UPDATE)).

The confirming SDP offer contains exactly the RTP payload type of the Selected Codec, as determined by the first SDP answer, e.g., in SIP 18x, see GSMA PRD IR.92 [2].

Note: Typically, also the RTP payload type of the telephone-event codec is included, but this is not used for encoding of media flows and not mentioned here any further.

Both the originating and the terminating IMS network must forward this RTP payload type of the Selected Code within the confirming SDP offer unmodified, unless transcoding or repacking is inserted.

Media flows may be subsequently modified within an existing SIP session via a new offer SDP carried within a SIP re-INVITE or SIP UPDATE message. Such a new SDP offer may include more RTP payload types for a potential modification of the media flows.

As stated in GSMA PRD IR.92 [2], SIP forking is recommended to be supported for inter-operability and forward-compatibility reasons, and the IBCF should be able to maintain at least forty (40) parallel early dialogues until receiving the final response on one of them. The IBCF/TrGW must support receiving media on one of these early dialogues as described in 3GPP TS 24.628 [67].

It is recommended that SDP offers and SDP answers are transited unchanged across the II-NNI to facilitate inter-operability and avoid transcoding. For voice, video and data channel, the mandatory and minimal content of the SDP offer and the content of the SDP answer is as described in section 3.2.1 and 2.4.3 of GSMA PRD IR.92 [2], section 3.3.1 of GSMA PRD IR.94 [3] and section 4.1 of GSMA PRD NG.134 [130] respectively, unless the SDP offer is modified by the originating network before reaching the II-NNI as specified below.

Some modifications of RTP (Real Time Protocol) payload types in the SDP offer by the originating network are permissible (except in a confirming SDP offer):

- The originating network can add additional RTP payload types describing codecs or codec configurations available via transcoding as less preferred alternatives (by placing those RTP payload types as last RTP payload types in the related SDP m-line), e.g. to increase the likelihood of interoperability.
- The originating network can remove some offered RTP payload types. However, the originating network must retain at least one AMR (Adaptive Multi-Rate) and one AMR-WB (AMR Wide Band) RTP payload type in the SDP offer. If the originating network and the subsequent II-NNI support super-wideband or fullband calls and such a call is offered, the originating network must also retain EVS RTP payload types as listed in section 2.4.3.3 of GSMA IR.92 [2].
- The originating network can restrict the configuration associated with an offered RTP payload type by modifying MIME parameters as allowed according to the SDP offer-answer rules specified for those MIME parameters (for AMR and AMR-WB see IETF RFC 4867 [93], for EVS see 3GPP Release 12 TS 26.445 [86], for H.264 see IETF RFC 6184 [94]). However, if a "mode-set" parameter is added for the AMR codec type, only value "mode-set=0,2,4,7" is recommended. If a "mode-set" parameter is added for the AMR-WB codec type, only value "mode-set=0,1,2" is recommended.
- Regardless, whether the MIME parameter "ch-aw-recv" is present or not in an EVS RTP payload type of the received SDP offer, the originating network can set the MIME parameter "ch-aw-recv" to any value allowed by TS 26.445 [86] in the forwarded SDP offer in an EVS RTP payload type.

- In the scenario of re-packing between AMR-WB and EVS IO mode at the II-NNI (see section 10.3.3.1), an EVS RTP payload type containing the MIME parameter value “evs-mode-switch=1” is here referred as an “EVO IO payload type.”
- The originating network can consistently replace an AMR-WB payload type with an EVS IO payload type in all SDP answers, as long as it also replaces that EVS IO payload type with the corresponding AMR-WB payload type in subsequent SDP offers. The originating network must not make such replacement in an initial SDP offer.

Similar modifications of RTP payload types in the received SDP offer are permissible for the terminating IMS network, before the final SDP offer is sent to the terminating client.

- If present in an EVS RTP payload type of the received SDP offer, the terminating network can keep the MIME parameter “ch-aw-recv” or increase its value, but it cannot decrease its value.
- If not present in an EVS RTP payload type of the received SDP offer, the terminating network can set the MIME parameter “ch-aw-recv” to any value allowed by 3GPP TS 26.445 [86], before the final SDP offer is sent to the terminating network.
- In the scenario of re-packaging between AMR-WB and EVS IO mode at II-NNI (see section 10.3.3.1), where an EVS RTP payload type contains the MIME parameter value “evs-mode-switch=1” is here referred as an “EVS IO payload type”:
  - The terminating network can, when the received initial SDP offer contains AMR-WB but no EVS payload types, add an EVS IO payload type with highest priority to the final SDP offer sent to the terminating client.
  - If such added EVS IO payload type in the final SDP offer was accepted by the terminating UE, the terminating network must consistently replace the corresponding AMR-WB payload type with the previously added EVS IO payload type and vice versa in any succeeding SDP offer/answer, as long as the received SDP contains only those corresponding speech payload types. Any subsequent received SDP offer containing more speech payload types should be handled as an initial SDP offer (see above).

## 10.2 RTP Profile

As stated in GSMA PRD IR.92 [2] and GSMA PRD IR.88 [6], the Real Time Protocol (RTP) profile and Audio Video Profile (RTP/AVP) (IETF RFC 3551 [32]) shall be used for voice sessions and Video sessions respectively.

As stated in GSMA PRD IR.94 [3], the Extended RTP Profile for Real Time Control Protocol (RTCP)-based Feedback (RTP/AVPF) (IETF RFC 4585 [33]) must be used for video telephony sessions. If the initial offer using RTP/AVPF is rejected, then a new offer shall be sent by the UE using RTP/AVP for the video telephony session.

If any RCS service is used, except for pager-mode messaging (which has no user plane), TCP/MSRP sessions shall be negotiated to support the required media plane flows as defined by IETF RFC 4975 [58] and described by GSMA PRD RCC.07 [44].

## 10.3 Codecs

The codecs in this profile are based on those mandated in the cited GSMA PRDs.

In general, whilst codec negotiation takes place end-end, the codecs supported at the NNI are subject to bi-lateral agreement. This profile defines a number of mandatory codecs at the NNI.

Further, this profile does not preclude other codecs being bilaterally agreed and nor should the IBCF unnecessarily remove codecs from a list received in an offer/answer as codec negotiation is an end-to-end negotiation.

Where possible, transcoding is to be avoided at the NNI. However, it is acknowledged that the IBCF/TrGW may need to perform transcoding at a given NNI where the received codec(s) are exclusive of a bi-laterally agreed codec list.

### 10.3.1 Audio Codecs

For this profile, the following audio codecs are mandatory, to be supported over the II-NNI, in line with GSMA PRD IR.92 [2]:

Adaptive Multi-Rate (AMR) codec (described in 3GPP TS 26.114 [9], 3GPP TS 26.071 [70], 3GPP TS 26.090 [71], 3GPP TS 26.073 [72], and 3GPP TS 26.104 [73]).

Adaptive Multi-Rate Wideband (AMR-WB) codec (described in 3GPP TS 26.114 [9], 3GPP TS 26.171 [75], 3GPP TS 26.190 [77], 3GPP TS 26.173 [76], 3GPP TS 26.193 [78] and 3GPP TS 26.204 [79]).

- If super-wideband or fullband speech communications are supported over the II-NNI, then the EVS codec is also mandatory as described in 3GPP Release 12 TS 26.114 [9], 3GPP Release 12 TS 26.441 [883], 3GPP Release 12 TS 26.445 [86], 3GPP Release 12 TS 26.442 [84], 3GPP Release 12 TS 26.443 [85], 3GPP Release 12 TS 26.447 [87], 3GPP Release 12 TS 26.449 [88], 3GPP Release 12 TS 26.450 [489] and 3GPP Release 12 TS 26.451 [90].
- The telephone-event codec according to IETF RFC 4733 [22], further refined in 3GPP TS 26.114 [9], with all relevant RTP clock rates.

An SDP offer over the II-NNI must contain at least the following:

- An AMR RTP payload type with no mode set specified or with mode-set=0,2,4,7, and/or an AMR-WB RTP payload type with no mode set specified or with mode-set=0,1,2.
- The telephone-event/8000 and/or the telephone-event/16000 RTP payload types, chosen such that a separate, matching payload type is included for every RTP clock rate used by speech codecs that are included in the same SDP offer.
- If a super-wideband or fullband call is offered and supported over the II-NNI based on bilateral agreement, EVS RTP payload types as listed in section 2.4.3.3 of IR.92 [2].

In addition, to support interoperability with non-3GPP access inter-connect, the following audio codecs are also recommended to be supported for this profile over the II-NNI :

G.711 (see IETF RFC 3551 [32]), using payload type 8 (A-law) and/or 0 (Mu-Law) dependent on market considerations.

Comfort Noise codec as specified in IETF RFC 3389 [95], for use with audio codecs lacking built-in comfort noise support, such as e.g. G.711.

Other audio codecs may be supported (e.g., G.729A and G.722 (see IETF RFC 3551 [32])) based on bilateral agreement and are out of scope of this profile.

### **10.3.2 Video Codecs**

Video codecs may be supported over the II-NNI based on agreement between operators. If video codecs are supported, the requirements in the present sub clause apply.

For this profile, the following video codecs and related profiles are mandatory to be supported over the II-NNI, in line with GSMA PRD IR.94 [3]:

ITU-T Recommendation H.264 Constrained High Profile (CHP) Level 3.1 as specified in 3GPP release 13 TS 26.114 [9] section 5.2.2,

ITU-T Recommendation H.264 Constrained Baseline Profile (CBP) Level 1.2 as specified in 3GPP Release 10 TS 26.114 [9] section 5.2.2,

ITU-T Recommendation H.264 Constrained Baseline Profile (CBP) Level 3.1 as specified in 3GPP Release 10 TS 26.114 [9] section 5.2.2.

For this profile, the following video codec and related profile is recommended to be supported over the II-NNI, in line with GSMA PRD IR.94 [3]

ITU-T Recommendation H.265 (HEVC) Main Profile, Main Tier, Level 3.1 as specified in 3GPP Release 12 TS 26.114 [9] section 5.2.2.

Other video codecs may be supported based on bilateral agreement and are out of scope of this profile.

### **10.3.3 Codec Negotiation/Handling at the NNI**

The network shall have the ability to perform transcoding between the codecs it supports within each media type. The network may implement these capabilities in various network elements; however, the remainder of this section assumes they are provided by the IBCF / TrGW.

To facilitate interoperability, the IBCF shall not preclude other codecs being offered across the NNI.

### **10.3.4 Re-packaging between AMR-WB and EVS IO modes**

In the following:

- AMR-WB IO mode is used as a synonym for AMR-WB interoperable mode (as an alternative implementation) using IETF RFC4867 (AMR-WB RTP) [93] payload format; and
- EVS IO mode is used as a synonym for AMR-WB interoperable mode using TS 26.445, Annex A (EVS RTP) [86] payload format.

The network may enforce usage of the EVS IO mode as replacement for AMR-WB by providing re-packaging the RTP payload at II-NNI. This relies on:

- SDP modification in SIP network nodes is feasible for the case when an EVS capable network is connected to a network without EVS capability but supporting AMR-WB.
- The SDP-modifying SIP network nodes are stateful, in the way they are capable to follow a SIP dialog and compare an SDP answer with the corresponding SDP offer.

Note: A UE as described in [2] supporting EVS and offering super-wideband or full-band speech communications is capable of EVS IO mode [9].

There are two ways to enforce the use of EVS in the EVS-supporting call leg:

- Using *EVS primary mode* in the EVS call leg and performing transcoding to AMR-WB in the network. The EVS call leg then benefits from improved error robustness, both in uplink and downlink. The improved EVS error robustness especially in channel aware mode allows to relax BLER target in that call leg which corresponds to an improved SNR and indoor/outdoor cell coverage.
- Using *EVS IO mode* and perform re-packaging in the network. This re-packaging procedure may be executed already during SRVCC if during an EVS VoLTE to VoLTE call, one UE switches to the CS network and AMR-WB codec. This also enables the use case where EVS IO mode is directly negotiated during call setup. It is likely advantageous with regard to error robustness and audio quality to use EVS IO mode instead of legacy AMR-WB.

Note: In case the AMR-WB codec is negotiated, the UEs may use the AMR-WB IO mode as the alternative implementation of AMR-WB for the sending and/or receiving direction. It is in the UE vendor's sole discretion to implement AMR-WB using AMR-WB IO mode. Hence, the operator/IMS network can only enforce the use of the EVS IO mode but not of AMR-WB IO.

Active Control and User Plane network nodes are assumed in the following, with IMS used as a synonym for all network nodes in the EPC and IMS involved in the call setup:

Originating UE <=> Originating IMS <=> Terminating IMS <=> Terminating UE

It is assumed that a UE supporting EVS can only be capable of using EVS if the network it is attached to also supports EVS. In all other cases, the UE is considered to be a non-EVS-capable UE.

The following SDP modifications in call signalling reflect the alternative connections enforcing the use of EVS IO mode:

An EVS-capable originating IMS (oIMS) that receives an SDP offer from the originating UE (oUE) containing one or more EVS payload types (oUE is EVS capable), and further receives the corresponding, confirming SDP answer from the terminating UE (tUE) not containing any EVS payload types (tUE is not EVS capable), replaces:

1. "AMR-WB" with "EVS IO mode" in this confirming answer and subsequent SDP answers sent towards the oUE (modification 1a), and

2. "EVS IO mode" with "AMR-WB" in subsequent SDP offers sent towards the tUE (modification 1b).

An EVS-capable terminating IMS (tIMS) that receives an initial SDP offer from the oIMS not containing any EVS payload types (oIMS is not EVS capable):

1. Adds an "EVS IO mode" payload type with highest priority to the final SDP offer forwarded to the tUE (modification 2a).
2. If tIMS receives a corresponding, confirming SDP answer from the tUE containing "EVS IO mode" (tUE is EVS capable), tIMS replaces:
  - "EVS IO mode" with "AMR-WB" in this confirming SDP answer and subsequent SDP answers sent towards oIMS (modification 2b), and
  - "AMR-WB" with "EVS IO mode" in any subsequent SDP offers sent towards tUE that contain only those speech payload types, e.g. in SIP UPDATE (modification 2c). Subsequent SDP offers containing more speech payload types should be handled as initial SDP offers (see above).

All modifications are assumed to be implemented in the IMS network instances already managing SDP messages, e.g. they may be implemented in P-CSCF or other IMS network instances such as the ATCF (Access Transfer Control Function) already present for SRVCC. The re-packaging may be implemented e.g. in Media Resource Function Processor (MRFP) or Access Transfer Gateways (ATGW).

SDP examples are provided in Annex B.1.

### **10.3.5 Global Text Telephony (GTT)**

If supported, Global Text Telephony (GTT) messages must use ITU-T Recommendation T.140 [24] real-time text according to the rules and procedures specified in 3GPP TS 26.114 [9] and with clarifications in Annex B.2 in GSMA PRD IR.92 [2].

### **10.3.6 DTMF**

DTMF (Dual Tone Multi Frequency) events shall be conveyed across the II-NNI via the "Named Telephone Event" payload format defined in IETF RFC 4733 [22], as specified in Annex G of 3GPP TS 26.114 [9], using the relevant telephone-event codec.

See 3GPP TS 26.114 [9] and clause 3.2 of GSMA PRD IR.92 [2] for further guidance on the encoding and transport of DTMF events.

### **10.3.6 IMS Data Channel**

If supported, IMS Data Channel media can be negotiated and conveyed across the NNI according to the rules and procedures specified in 3GPP TS 26.114 [9], 3GPP TS 29.165 [1] and with clarifications in GSMA PRD NG.134 [130].

## **10.4 Early Media Detection**

Early media shall be supported in this profile.

As stated in GSMA PRD IR.92 [2], early media may be indicated by the presence of a P-Early-Media header in a re-INVITE/UPDATE/PRACK request or a 18x (INVITE) response. A



P-Early-Media shall be transited across the NNI and all values of the P-Early-Media header shall be supported at the NNI. In addition, the IBCF shall configure its TrGW in a manner consistent with the P-Early-Media header to ensure that the related media traverses the TrGW.

## 10.5 SDP Contents

SDP is defined in IETF RFC 4566 [13].

SDP usage shall be compliant with the Offer/Answer rules in IETF RFC 4566 [13]

NAPT may be performed at the NNI in the user plane via the TrGW under the control of the IBCF via the Ix reference point as defined in 3GPP TS 29.238 [17].

If NA(P)T is applied at the NNI, then the IP address and port in the c=line and m=line respectively must be modified.

For handling MSRP media, Border Elements shall follow GSMA PRD IR.90 [5]. In the case of SIMPLE IM, if NA(P)T is performed, then the IP address and port must also be modified in the a=path attribute, whilst leaving the session identity unchanged. For a summary of the different procedures for MSRP handling, see also section 5.19 of 3GPP TS 23.334 [92].

The SDP contents applicable to the MMTel and RCS services in this document are summarized in Table 8 below.

SDP Attribute	Profile Settings
Version (v-line)	The value must always be equal to zero: v=0.
Origin (o-line)	The origin line consists of six fields: ( <code>&lt;username&gt;</code> , <code>&lt;sess-id&gt;</code> , <code>&lt;sess-version&gt;</code> , <code>&lt;nettype&gt;</code> , <code>&lt;addrtype&gt;</code> and <code>&lt;unicast-address&gt;</code> ).  - <code>&lt;user name&gt;</code> should contain an hyphen - <code>&lt;session ID&gt;</code> and <code>&lt;version&gt;</code> should contain one or more digits as described in IETF RFC 4566 [13] - <code>&lt;network type&gt;</code> shall be set to IN - <code>&lt;address type&gt;</code> shall be set to IP4 or IP6 The Address Type shall be set to "IP4" or "IP6" depending on the addressing scheme. - <code>&lt;address&gt;</code> should contain the fully qualified domain name or IP address of the media end point (typically the TrGW).  Example: "o=- 0 0 IN IP4 10.1.2.3"
Session (s-line)	The session name "s=" line contains a single field s= <code>&lt;session name&gt;</code> .  Example: "s=-"
Time	The time "t=" line consists of two fields

SDP Attribute	Profile Settings
(t-line)	<p>t= &lt;start time&gt; and &lt;stop time&gt;.</p> <p>Example:                      "t=0 0"</p>
Connection (c-line)	<p>The connection "c=" line consists of three fields,                      c=&lt;nettype&gt; &lt;addr type&gt; &lt;connection-address&gt;                      The &lt;nettype&gt; shall be "IN"                      The &lt;addr type&gt; shall be "IP4" or "IP6"                      The &lt;connection-address&gt; shall be an IPv4 or IPv6 address as signified by the                      &lt;addr type&gt;.</p> <p>Example:                      "c=IN IP4 10.10.1.1"</p>
Media (m-line)	<p>The media "m=" line consists of four fields &lt;media&gt;, &lt;port&gt;, &lt;proto&gt; and &lt;fmt&gt;                      The &lt;media&gt; field shall be set to "audio" or "video" or "message" or "text" or                      "image" or "application".</p> <p>The &lt;port&gt; is set to the port number that shall send/receive the media. The                      port number must be even for audio/video/text/image.</p> <p>The &lt;proto&gt; is set to "RTP/AVPF" for video and to "RTP/AVP" for                      audio/video/text media and to "TCP/MSRP" for message media.</p> <p>The &lt;fmt&gt; is set to one or more RTP payload type numbers for RTP/AVP and                      RTP/AVPF and to "*" for TCP/MSRP.</p> <p>Examples:                      "m=audio 1234 RTP/AVP 100 8 0"                      "m=video 1234 RTP/AVPF 100 102"                      "m=video 1234 RTP/AVP 100 102"                      "m=message 1239 TCP/MSRP *"                      "m=text 1234 RTP/AVP 99"                      "m=application 1234 UDP/DTLS/SCTP webrtc-datachannel"</p>
Media Attributes (a-lines)	<p>The attribute "a=" line consists of one or two fields &lt;attribute&gt; or                      &lt;attribute&gt;:&lt;value&gt;. Many attribute lines are defined; of which most are related                      to a specific payload type.</p> <p>One or more media attribute lines may be included dependent on the payload                      type. In general, attribute lines should be transited at the NNI to facilitate media                      interworking. Some common media attribute lines are shown below:</p> <p>(for stream activity)                      a=inactive/recvonly/sendonly/sendrecv                      Example: "a=inactive"</p> <p>(for identifying RTP payload type numbers and mapping them to codec types)</p>

SDP Attribute	Profile Settings
	<p>a= rtpmap: &lt;payload type number&gt; &lt;encoding name&gt;/&lt;clock rate&gt; [/&lt;encoding parameters&gt;]                      Example: "a=rtpmap:100 AMR-WB/16000"</p> <p>(for providing payload type specific parameters)                      a= fmp: &lt;format&gt; &lt;format specific parameters&gt;                      Example: "a=fmp:100 mode-change-capability=2"                      Note: When m= line profile is (S)RTP/AVP(F), "format" is always an RTP payload type number</p> <p>(for defining packetization time in ms)                      a= ptime: &lt;time&gt;                      Example: a=ptime:20</p> <p>(for segmented QOS indication as per IETF RFC 3312 [15])                      a=curr: &lt;precondition type&gt; &lt;status-type&gt; &lt;direction&gt;                      a=des: &lt;precondition type&gt; &lt;strength&gt; &lt;status-type&gt; &lt;direction&gt;                      a=conf: &lt;precondition type&gt; &lt;status-type&gt; &lt;direction&gt;                      Example :                      a=curr:qos local none                      a=des:qos mandatory local sendrecv                      a=curr:qos remote none                      a=des:qos optional remote sendrecv                      a=conf:qos remote sendrecv</p> <p>(for OMR as per 3GPP TS 29.079 [37])                      a=visited-realm: &lt;instance&gt; &lt;realm&gt; &lt;addrtype&gt; &lt;addr&gt; &lt;port&gt;                      a=omr-m-cksum: &lt;hexNumber&gt;                      a=omr-s-cksum: &lt;hexNumber&gt;                      a=omr-codecs: &lt;instance&gt;: &lt;proto&gt; 1* &lt;codec&gt;                      a=omr-m-att: &lt;instance&gt;: 1* &lt;attribute&gt;                      a=omr-s-att: &lt;instance&gt;: 1* &lt;attribute&gt;                      a=omr-m-bw: &lt;instance&gt;: &lt;bandwidth&gt;                      a=omr-s-bw: &lt;instance&gt;: &lt;bandwidth&gt;                      Example:                      a=visited-realm:1 Xa.operatorX.net IN IP4 192.0.2.1 49170                      a=visited-realm:2 XY.operator.netX IN IP4 13.24.1.1 66000                      a=omr-m-cksum:89                      a=omr-s-cksum:0</p> <p>(for RTP Profile Negotiation as per IETF RFC 5939 [35])                      a=tcap: &lt;capability number&gt; &lt;transport protocol list&gt;                      a=pcfg: &lt;config number&gt; &lt;potential config list&gt;                      a=acfg: &lt;config number&gt; &lt;selected config list&gt;</p>

SDP Attribute	Profile Settings
	<p>Example:  a=tcap:1 RTP/AVPF  a=pcfg:1 t=1</p> <p>(for RTCP Feedback as per IETF RFC 4585 [33] and IETF RFC 5104 [100])  a=rtcp-fb:&lt;payload type&gt; &lt;feedback type&gt; &lt;feedback parameters&gt;  Example:  a=rtcp-fb:* nack rpsi  a=rtcp-fb:* ccm fir  a=rtcp-fb:* ccm tmmb</p> <p>(for RTP Header Extensions as per IETF RFC 5285 [101])  a=extmap:&lt;ext id&gt; &lt;extension attributes&gt;  Example:  a=extmap:7 urns:3gpp: video-orientation</p> <p>(for TCP Connection establishment for MSRP as per IETF RFC 4975 [58])  a=path:msrp://10.10.1.1:1239/jshA7weztas;tcp</p> <p>(for supporting the presentation of an authentication certificate as part of the TLS handshake procedure during the establishment of an IMS Data Channel as per IETF RFC 4572 [134])  a=fingerprint:&lt;hash-func&gt; &lt;certificateID&gt;  Example:  a=fingerprint:SHA-1  4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:AB</p> <p>(to indicate maximum SCTP user message size (indicated in bytes) that an SCTP endpoint is willing to receive on the SCTP association associated with the "m=" line as per IETF RFC 8841 [131])  a=max-message-size: &lt;maximum-byte-number&gt;  Example:  a=max-message-size: 1024</p> <p>(to indicate the SCTP port number associated with an SDP media description ("m=" line) with a "UDP/DTLS/SCTP" or a "TCP/DTLS/SCTP" protocol value as described in IETF RFC 8841 [131])  a=sctp-port: &lt;port&gt;  Example:  a=sctp-port:5000</p> <p>(to indicate which of the end points should initiate the TCP connection establishment (i.e., send the initial TCP SYN) as described in IETF RFC 4145 [135])  a=setup: &lt;role&gt;  Example:</p>

SDP Attribute	Profile Settings
	<p>a=setup:actpass</p> <p>(to uniquely identifies the DTLS association or TLS connection via the attribute values of the offerer and the answerer as described in IETF RFC 8842 [132])</p> <p>a=tls-id:&lt;TLS-Attrubute-Values&gt;</p> <p>Example:</p> <p>a=tls-id: abc3de65cddef001be82</p> <p>(to define data-channel-specific parameters as described in IETF RFC 8864 [133])</p> <p>a=dcmap:&lt;Data-Channel-Parameters&gt;</p> <p>Example:a=dcmap:0 subprotocol="http"</p> <p>(to define attributes that are specific to data-channel sub-protocols as described in IETF RFC 8864 [133])</p> <p>a=dcsa:&lt;Data-Channel-Sub-Protocol-Arrtibutes&gt;</p> <p>Example:a=dcsa ??</p> <p>(to indicate the UE's desired QoS properties from the local access network as described in 3GPP TS 26.114 [9])</p> <p>a=3gpp-qos-hint: 1*&lt;qos-hint-property&gt;</p> <p>Example:</p> <p>a=3gpp-qos-hint:loss=0.00001;latency=300</p>
Bandwidth (b-line)	<p>The bandwidth "b=" line consists two fields &lt;bwtype&gt;:&lt;bandwidth&gt;.</p> <p>For voice and video call sessions, as mandated in GSMA PRDs IR.92 [2] and IR.94 [3] the &lt;bwtype&gt; is set to AS for the RTP media part, and to RS / RR for RTCP.</p> <p>For GTT sessions, as mandated in GSMA PRD IR.92 [2], the &lt;bwtype&gt; is set to AS, with RS and RR set to zero.</p> <p>For RCS sessions, the &lt;bwtype&gt; is set to AS.</p> <p>The &lt;bandwidth&gt; defines the peak bandwidth in units of kbits/sec (for AS) or bits/sec (for RS/RR) and includes up to and including the IP layer as defined in IETF RFC 4566 [13] and IETF RFC 3566 [18].</p> <p>Examples:                      "b=RR:100" , "b=AS:100"</p>

**Table 8: SDP Contents**

## 10.6 RTP/RTCP Packet Source

In a SIP session, the same IP address and port number shall be used to send and receive RTP packets (symmetric IP address and port number). Further, the port number for sending/receiving RTCP packets shall be equal to "the port number negotiated for RTP" + 1.

SDP Bandwidth Modifiers for RTCP (see IETF RFC 3556 [18]) are supported on the NNI.

## 11 IP Version

As stated in 3GPP TS 29.165 [1], the supported IP version at the NNI may be IPv4 only, IPv6 only or both IPv4 and IPv6.

In case IPv4 only and IPv6 only networks are interconnected, the involved IBCF and TrGWs shall apply the IP version interworking procedures as indicated in 3GPP TS 29.162 [69].

## 12 Inter-Operator Accounting

Inter-operator accounting shall be performed as described in section 11.2 of 3GPP TS 29.165 [1].

## 13 Inter-working between different IMS Core Network Deployments

When there is a mismatch of IMS core network deployment options between Operators, then suitable bi-lateral agreements must be in place at the NNI to ensure consistency of message routing for all services. Such a bi-lateral agreement must define whether:

- The NNI is applicable to a single IMS core network supporting all services defined in section 2, or
- The NNI is applicable to one of two separate IMS core networks supporting only a sub-set of services as defined in section 2.

As stated in section 5.7 of GSMA PRD IR.65 [4], there is a need to provide inter-connect between MNOs that have chosen different deployment options (i.e. single IMS core versus two separate IMS core networks). In this case there is a need to be able to:

Resolve a single E.164 Public User Identity to multiple SIP URIs (one for the IMS core providing MMTEL services and one for the IMS core providing RCS services) as described in section 4.2.3 of GSMA PRD NG.105 [97],

Route messages correctly from the single IMS core network to one or both of the separate IMS core networks depending on the context of the related request (i.e. MMTEL related, RCS related or both).

Note: A capability exchange from a single IMS core network must be sent to both of the separate IMS core networks to enable the full set of services (i.e. including both MMTEL and RCS services) to be returned to the single IMS core network.

Details of the interworking between different IMS core network deployments are provided in annex C with example message flows in annex E.

## 14 Inter-working between Capability Exchange Methods

Operators can be misaligned in terms of their preferred mechanism to perform a capability exchange (i.e. Options based versus Presence based). In this case, the bi-lateral agreement at a given NNI must also specify the capability exchange mechanism that is applicable at

that NNI. Interworking between Options and Presence based capability exchange is described in GSMA PRD IR.90 [5] section 4.1.3. Such an interworking function can be provided by either network or via an IPX provider.

Details of the interworking between capability exchange methods is provided in annex D with example message flows in annex E.

## Annex A SIP Header Examples (Informational)

Table 9 below provides examples and references to the relevant RFCs etc. for the SIP headers in scope in this profile.

Header	Reference(s)	Example Header
Accept	IETF RFC 3261 [12]	<i>Accept:application/sdp,message/cpim</i>
Accept-Contact	IETF RFC 3841 [99]	<i>Accept-Contact:*,mobility="mobile";methods="INVITE"</i>
Accept-Encoding	IETF RFC 3261 [12]	<i>Accept-Encoding:gzip</i>
Accept-Language	IETF RFC 3261 [12]	<i>Accept-Language: da, en-gb;q=0.8, en;q=0.7</i>
Accept-Resource-Priority	IETF RFC 4412 [66]	<i>Accept-Resource-Priority:ets.0</i>
Alert-Info	IETF RFC 3261 [12]	<i>Alert-Info:&lt;urn:alert:service:call-waiting&gt;</i>
Allow	IETF RFC 3261 [12]	<i>Allow:INVITE, ACK, UPDATE, PRACK, CANCEL, PUBLISH, MESSAGE, OPTIONS,SUBSCRIBE,NOTIFY</i>
Allow-Events	IETF RFC 6665 [102]	<i>Allow-Events: conference</i>
Answer-Mode	IETF RFC 5373 [103]	<i>Answer-Mode: Auto</i>
Authentication-Info	IETF RFC 3261 [12] IETF RFC 3310 [104]	<i>Authentication-Info: nextnonce="47364c23432d2e131a5fb210812c"</i>
Authorization	IETF RFC 3261 [12] IETF RFC 3310 [104] 3GPP TS 24.229 [68]	<i>Authorization: Digest, username="jon.dough@mobile.biz", realm="RoamingUsers@mobile.biz", nonce="CjPk9mRqNuT25eRkajM09uTI9nM09uTI9nMz5OX25PZz==", uri="sip:home.mobile.biz", qop=auth-int, nc=00000001,cnonce="0a4f113b", response="6629fae49393a05397450978507c4ef1", opaque="5ccc069c403ebaf9f0171e9517f40e41"</i>
Call-Id	IETF RFC 3261 [12]	<i>Call-Id:12345@mydomain.com</i>
Call-Info	IETF RFC 3261 [12]	<i>Call-Info: &lt;http://www.example.com/alice/photo.jpg&gt;;purpose=icon</i>
Contact	IETF RFC 3261 [12]	<i>Contact:&lt;sip:ibcf4@operator1.com:5060&gt;</i>
Content-Disposition	IETF RFC 3261 [12]	<i>Content-Disposition:session</i>



Header	Reference(s)	Example Header
Content-Encoding	IETF RFC 3261 [12]	<i>Content-Encoding:gzip</i>
Content-Language	IETF RFC 3261 [12]	<i>Content-Language:fr</i>
Content-Length	IETF RFC 3261 [12]	<i>Content-Length:146</i>
Content-Type	IETF RFC 3261 [12]	<i>Content-Type:application/sdp</i>
Contribution-ID	OMA CPM [34]	<i>Contribution-ID: abcdef-1234-5678-90ab-cdef01234567</i>
Conversation-ID	OMA CPM	<i>Conversation-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6</i>
CSeq	IETF RFC 3261 [12]	<i>CSeq:2 CANCEL</i>
Date	IETF RFC 3261 [12]	<i>Date Sat, 13 Nov 2010 23:29:00 GMT</i>
Error-Info	IETF RFC 3261 [12]	<i>Error-Info: &lt;sip:announcement10@example.com&gt;</i>
Event	IETF RFC 6665 [102] IETF RFC 6446 [105] IETF RFC 3680 [50]	<i>Event:reg</i>
Expires	IETF RFC 3261 [12]	<i>Expires:3600</i>
Feature-Caps	IETF RFC 6809 [106] 3GPP TS 24.229 [68]	<i>Feature-Caps:+3gpp.trf=sip:trf3.operator3.com</i>
Flow-Timer	IETF RFC 5626 [107]	<i>Flow-Timer: 30</i>
From	IETF RFC 3261 [12]	<i>From: &lt;sip:+12125551212@operator1.com;user=phone&gt;;tag=145688</i>
Geolocation	IETF RFC 6442 [108]	<i>Geolocation:&lt;user1@operator1.com&gt;</i>
Geolocation-Error	IETF RFC 6442 [108]	<i>Geolocation-Error:100;code="Cannot Process Location"</i>
Geolocation-Routing	IETF RFC 6442 [108]	<i>Geolocation-Routing:yes</i>
History-Info	IETF RFC 4244	<i>History-Info: &lt;sip:Bob@P1.example.com&gt;;index=1, &lt;sip:Bob@P2.example.com&gt;; index=1.1, &lt;sip:User3@UA3.example.com? Reason=SIP%3Bcause%3D486%3Btext%3D"Busy Here"&gt;;index=1.2,</i>

Header	Reference(s)	Example Header
		<sip:User5@UA5.example.com>;index=1.3
Info-Package	IETF RFC 6086 [82]	Info-Package:foo
InReplyTo-Contribution-ID	OMA CPM [34]	InReplyTo-Contribution-ID: 01234567-89ab-cdef-0123-456789abcdef
Join	IETF RFC 3911 [109]	Join: 1234@example.com;to-tag=3456;from-tag=6789
Max-Breadth	IETF RFC 5393 [110]	Max-Breadth:20
Max-Forwards	IETF RFC 3261 [12]	Max-Forwards:70
Message-Expires	OMA CPM [34]	Message-Expires: 259200
Message-UID	OMA CPM [34]	Message-UID: 4392
MIME-Version	IETF RFC 3261 [12]	MIME-Version:1.0
Min-Expires	IETF RFC 3261 [12]	Min-Expires:40
Min-SE	IETF RFC 4028 [16]	Min-SE:60
Organization	IETF RFC 3261 [12]	Organization:My Company
P-Access-Network-Info	IETF RFC 7315 [111]	P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
P-Answer-State	IETF RFC 4964 [112]	P-Answer-State: Unconfirmed
P-Asserted-Identity	IETF RFC 3325 [113]	P-Asserted-Identity: sip:+14085264000@operator1.com;user=phone
P-Asserted-Service	IETF RFC 6050 [114]	P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
P-Associated-URI	IETF RFC 7315 [111]	P-Associated-URI:sip:user2@operator3.com
P-Called-Party-ID	IETF RFC 7315 [111]	P-Called-Party-ID: sip:user1-business@example.com
P-Charging-Vector	IETF RFC 7315 [111]	P-Charging-Vector: icid-value=1234bc9876e; icid-generated-at=192.0.6.8; orig-ioi=home1.net
P-Early-Media	IETF RFC 5009 [115]	P-Early-Media:supported

Header	Reference(s)	Example Header
P-Preferred-Service	IETF RFC 6050 [114]	<i>P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel</i>
P-Private-Network-Indication	IETF RFC 7316 [116]	<i>P-Private-Network-Indication: example.com</i>
P-Profile-Key	IETF RFC 5002 [117]	<i>P-Profile-Key:sip:user3@operator4.com</i>
P-Refused-URI-List	IETF RFC 5318 [118]	<i>P-Refused-URI-List: sip:friends-list@example.net; members=&lt;cid:an3bt8jf03@example.net&gt;</i>
P-Served-User	IETF RFC 5502 [119]	<i>P-Served-User:sip:user1@operator2.com;sescase=orig</i>
P-Visited-Network-ID	IETF RFC 7315 [111]	<i>P-Visited-Network-ID:"Visited network number 1"</i>
Path	IETF RFC 3327 [20]	<i>Path:&lt;sip:P2.example.com;lr&gt;,&lt;sip:P1.example.com;lr&gt;</i>
Permission-Missing	IETF RFC 5360 [120]	<i>Permission-Missing: userC@example.com</i>
Policy-Contact	IETF RFC 6794 [121]	<i>Policy-Contact: sip:server5@example.com</i>
Priority	IETF RFC 3261 [12]	<i>Priority: emergency</i>
Priv-Answer-Mode	IETF RFC 5373 [103]	<i>Priv-Answer-Mode: Auto</i>
Privacy	IETF RFC 3323 [122] IETF RFC 3325 [113]	<i>Privacy:user</i> <i>Privacy:id</i>
Proxy-Authenticate	IETF RFC 3261 [12]	<i>Proxy-Authenticate: Digest realm="atlanta.com", domain="sip:ss1.carrier.com", qop="auth", nonce="f84f1cec41e6cbe5aea9c8e88d359", opaque="", stale=FALSE, algorithm=MD5</i>
Proxy-Authorization	IETF RFC 3261 [12]	<i>Proxy-Authorization: Digest username="Alice", realm="atlanta.com", nonce="c60f3082ee1212b402a21831ae", response="245f23415f11432b3434341c022"</i>
Proxy-Require	IETF RFC 3261 [12]	<i>Proxy-Require: foo</i>
Rack	IETF RFC 3262 [14]	<i>Rack:10</i>
Reason	IETF RFC 3326 [123]	<i>Reason:Q.850;cause=16;text="Terminated"</i>
Record-Route	IETF RFC 3261 [12]	<i>Record-Route: &lt;sip:server10.biloxi.com;lr&gt;,&lt;sip:bigbox3.site3.atlanta.com;lr&gt;</i>

Header	Reference(s)	Example Header
Recv-Info	IETF RFC 6086 [82]	<i>Recv-Info: bar</i>
Referred-By	IETF RFC 3892 [124]	<i>Referred-By: &lt;sip:alice@phone1.example.org&gt;</i>
Refer-Sub	IETF RFC 4488 [21]	<i>Refer-Sub=false</i>
Refer-To	IETF RFC 3515 [51]	<i>Refer-To: &lt;sip:dave@bobster.example.org? Replaces=425928%40bobster.example.com.3%3B to-tag%3D7743%3Bfrom-tag%3D6472&gt;</i>
Reject-Contact	IETF RFC 3841 [99]	<i>Reject-Contact: *,actor="msg-taker";video</i>
Replaces	IETF RFC 3891 [19]	<i>Replaces: 425928@bobster.example.org;to-tag=7743;from-tag=6472</i>
Reply-To	IETF RFC 3261 [12]	<i>Reply-To: Bob &lt;sip:user4@example.com&gt;</i>
Request-Disposition	IETF RFC 3841 [99]	<i>Request-Disposition: proxy</i>
Require	IETF RFC 3261 [12]	<i>Require:Path</i>
Resource-Priority	IETF RFC 4412 [66]	<i>Resource-Priority:ets.1</i>
Retry-After	IETF RFC 3261 [12]	<i>Retry-After: 3600</i>
Route	IETF RFC 3261 [12]	<i>Route: &lt;sip:bigbox3.site3.atlanta.com;lr&gt;, &lt;sip:server10.biloxi.com;lr&gt;</i>
RSeq	IETF RFC 3262 [14]	<i>RSeq:10</i>
Security-Client	IETF RFC 3329 [125]	<i>Security-Client: ipsec-ike</i>
Security-Server	IETF RFC 3329 [125]	<i>Security-Server: ipsec-ike</i>
Security-Verify	IETF RFC 3329 [125]	<i>Security-Verify:ipsec-ike; q=0.1</i>
Server	IETF RFC 3261 [12] & GSMA PRD IR.92 [2]	<i>Server: PRD-IR92/11 term-Vendor1/Model1-XXXX device-type/feature-phone mno-custom/none</i>
Service-Route	IETF RFC 3608 [126]	<i>Service-Route:&lt;sip:P1.example.com;lr&gt;,&lt;sip:P2.example.com;lr&gt;</i>
Session-Expires	IETF RFC 4028 [16]	<i>Session-Expires:3600;refresher=uac</i>
Session-Replaces	OMA CPM [34]	<i>Session-Replaces: abcdef-1234-5678-90ab-cdef01234567</i>

Header	Reference(s)	Example Header
Session-ID	IETF RFC 7329 [81]	<i>Session-ID: 0123456789abcdef123456789abcdef0</i>
SIP-Etag	IETF RFC 3903 [127]	<i>SIP-Etag:12345</i>
SIP-If-Match	IETF RFC 3903 [127]	<i>SIP-If-Match:12345</i>
Subscription-State	IETF RFC 6665 [102]	<i>Subscription-State:active</i>
Supported	IETF RFC 3261 [12]	<i>Supported:100Rel,timer,precondition</i>
Suppress-If-Match	IETF RFC 5839 [128]	<i>Suppress-If-Match:34567</i>
Target-Dialog	IETF RFC 4538 [129]	<i>Target-Dialog:</i>
Timestamp	IETF RFC 3261 [12]	<i>Timestamp:64</i>
To	IETF RFC 3261 [12]	<i>To: &lt;sip:+12126661212@operator2.com;user=phone&gt;;tag=1234s56</i>
Trigger-Consent	IETF RFC 5360 [120]	<i>Trigger-Consent: sip:user1@example.com; target-uri="sip:user2@example.xcom"</i>
Unsupported	IETF RFC 3261 [12]	<i>Unsupported:100Rel</i>
User-Agent	IETF RFC 3261 [12] & GSMA PRD IR.92 [2]	<i>User-Agent: PRD-IR92/11 term-Vendor1/Model1-XXXX device-type/feature-phone mno-custom/none</i>
Via	IETF RFC 3261 [12]	<i>Via: SIP/2.0/UDP operator1.com:5060;branch=z9hG4bK87asdks7</i>
Warning	IETF RFC 3261 [12]	<i>Warning: 307 isi.edu "Session parameter 'foo' not understood"</i>
WWW-Authenticate	IETF RFC 3261 [12] IETF RFC 3310 [104] 3GPP TS 24.229 [68]	<i>WWW-Authenticate: Digest realm="atlanta.com", domain="sip:boxesbybob.com", qop="auth", nonce="f84f1cec41e6cbe5aea9c8e88d359", opaque="", stale=FALSE, algorithm=MD5</i>

**Table 9: SIP Header Examples**

## **Annex B Message Examples (Informational)**

Annex B provides informational example message flows and message contents across the NNI.

The following topics are covered: -

- Re-Packaging between AMR-WB and EVS IO Modes.

### **B.1 Re-packaging between AMR-WB and EVS IO modes**

The SDP modifications during the offer-answer along with the relevant parts of SIP commands are described below.

Note: The message sequence is reduced to the codec negotiation relevant messages. Also note that SDP messages are shortened to improve readability.

### **B.2 Example 1: oUE + oIMS are EVS and AMR-WB capable, tUE + tIMS are only AMR-WB capable**

	Originating UE EVS +AMR-WB capable	Originating IMS EVS+ AMR- WB capable	Terminating IMS AMR-WB capable	Terminating UE AMR-WB capable
<b>SIP Invite</b>	Initial SDP offer ----- → m=audio 3000 RTP/AVP 96 97 a=rtpmap:96 EVS/16000 a=fmtp:96 br=5.9-24.4;bw=nb-swb a=rtpmap:97 AMR-WB/16000 a=fmtp:97 mode-set=0,1,2			
<b>SIP 183 Progress</b>			←----- Confirming SDP answer m=audio 3000 RTP/AVP 97 a=rtpmap:97 AMR-WB/16000 a=fmtp:97 mode-set=0,1,2	
	←----- Confirming SDP answer (modification 1a – replace AMR-WB with EVS IO mode) m=audio 3000 RTP/AVP 96 a=rtpmap:96 EVS/16000 a=fmtp:96 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1			
<b>SIP UPDATE</b>	Subsequent SDP offer ----- -→ m=audio 3000 RTP/AVP 96 a=rtpmap:96 EVS/16000 a=fmtp:96 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1			
			Subsequent SDP offer ----- → (modification 1b – replace EVS IO mode with AMR-WB) m=audio 3000 RTP/AVP 97 a=rtpmap:97 AMR-WB/16000 a=fmtp:97 mode-set=0,1,2	
<b>SIP 200 OK</b>	←----- Subsequent SDP answer m=audio 3000 RTP/AVP 97 a=rtpmap:97 AMR-WB/16000 a=fmtp:97 mode-set=0,1,2			

	<p>←----- Subsequent SDP answer (modification 1a – replace AMR-WB with EVS IO mode) m=audio 3000 RTP/AVP 96 a=rtpmap:96 EVS/16000 a=fmtp:96 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1</p>	
--	---	--

Applied modifications:

- modification 1a: originating IMS network modifies received confirming and subsequent SDP answer by replacing AMR-WB with EVS IO mode (in the sent SDP answer)
- modification 1b: originating IMS network replaces EVS IO mode in received subsequent SDP offer with AMR-WB (in sent subsequent SDP offer)

With those SDP modifications, EVS IO mode is used on originating UE, while AMR-WB is used on terminating UE. The originating IMS network needs to perform RTP re-packaging to connect the two UEs, but transcoding is not necessary. In the example above, RTP re-packaging would include translating RTP payload type 96 in the RTP header received from the originating UE to RTP payload type 97 in the RTP header sent towards the terminating IMS network, and vice versa.



**B.3 Example 2: oUE + oIMS are only AMR-WB capable, tUE + tIMS are EVS and AMR-WB capable**

	Originating UE AMR-WB capable	Originating IMS AMR-WB capable	Terminating IMS EVS + AMR-WB capable	Terminating UE EVS + AMR-WB capable
<b>SIP Invite</b>	Initial SDP offer ----->			
	m=audio 3000 RTP/AVP 96 a=rtpmap:96 AMR-WB/16000 a=fmtp:96 mode-set=0,1,2			
			Final SDP offer ----- -> (modification 2a – probe for EVS support) m=audio 3000 RTP/AVP 97 96 a=rtpmap:97 EVS/16000 a=fmtp:97 br=5.9-24.4;bw=nb-swb; mode- set=0,1,2;evs-mode-switch=1 a=rtpmap:96 AMR-WB/16000 a=fmtp:96 mode-set=0,1,2	
<b>SIP 183 Progress</b>			<----- Confirming SDP answer m=audio 3000 RTP/AVP 97 a=rtpmap:97 EVS/16000 a=fmtp:97 br=5.9-24.4;bw=nb-swb; mode- set=0,1,2;evs-mode-switch=1	
		<----- Confirming SDP answer (modification 2b – replace EVS IO mode with AMR-WB) m=audio 3000 RTP/AVP 96 a=rtpmap:96 AMR-WB/16000 a=fmtp:96 mode-set=0,1,2		
<b>SIP UPDATE</b>	Subsequent SDP offer ----- ->			
	m=audio 3000 RTP/AVP 96 a=rtpmap:96 AMR-WB/16000 a=fmtp:96 mode-set=0,1,2			
			Subsequent SDP offer ----- -> (modification 2c – replace AMR-WB with EVS IO mode) m=audio 3000 RTP/AVP 97 a=rtpmap:97 EVS/16000 a=fmtp:97 br=5.9-24.4; bw=nb-swb; mode- set=0,1,2; evs-mode-switch=1	

<b>SIP 200 OK</b>	←----- Subsequent SDP answer m=audio 3000 RTP/AVP 97 a=rtpmap:97 EVS/16000 a=fmtp:97 br=5.9-24.4.2;bw=nb-swb, mode-set=0,1,2;evs-mode-switch=1
	Subsequent SDP answer ←----- ---- (modification 2b – replace EVS IO mode with AMR-WB) m=audio 3000 RTP/AVP 96 a=rtpmap:96 AMR-WB/16000 a=fmtp:96 mode-set=0,1,2

Applied modifications:

- modification 2a: terminating IMS network adds EVS IO mode with highest priority to received SDP initial offer to probe terminating UE for EVS (in final SDP offer)
- modification 2b: terminating IMS network replaces EVS IO mode in received confirming or subsequent SDP answer with AMR-WB (in the sent SDP answer)
- modification 2c: terminating IMS network replaces AMR-WB with EVS IO mode (in subsequent SDP offer)

With those SDP modifications, EVS IO mode is used on terminating UE, while AMR-WB is used on originating UE. The terminating IMS network needs to perform RTP re-packaging to connect the two UEs, but transcoding is not necessary. In the example above, RTP re-packaging would include translating RTP payload type 96 in the RTP header received from the originating IMS network to RTP payload type 97 in the RTP header sent towards the terminating UE, and vice versa.

## **Annex C Interworking between IMS Core Network Deployments (Normative)**

As stated in section 5.7 of GSMA PRD IR.65 [4], there is a need to provide inter-connect between MNOs that have chosen different deployment options (i.e. single converged IMS core versus two separate IMS core networks). These deployment options are related to the different UE registration procedures defined in GSMA PRD NG.102 [10]. This annex addresses how to interwork the NNIs between the different deployment options of a converged IMS core network to a separated IMS core.

**Note:** The registration procedures in GSMA PRD NG.102 [10] allow a “single registration device” which supports a single IMS registration to a converged IMS core network as well as a “dual registration device” which supports two separate IMS registrations, either to a single IMS core network or to two separate IMS core networks (with one network providing MMTEL services and the other network providing RCS services). The deployment configuration supported by the device is configurable by the MNO.

To enable interworking between a single registration device and a dual registration device, there are two challenges: -

1. To discover the IMS communication services supported by another device, a single registration device sends a single UE Capability Exchange (UCE) request. It expects to receive in the UCE response service tuples associated with both MMTEL and RCS services in a single response. Unless the interworking and aggregation defined in this annex is used, the response will contain one or the other but not both.
2. A request sent to a dual registration device must reach the client or network element associated with the requested service. When the dual registration device is registered to separate IMS core networks, there are two different IMS network addresses serving the device.

To address these challenges, the following functionality is required to be provided by the network:-

1. Enhancements to ENUM to determine whether a single telephone number resolves to one or two SIP URIs as described in section 4.2.3 of GSMA PRD NG.105 [97]. A single SIP URI indicates a single IMS core network providing both MMTEL and RCS services and applicable to a single registration device or a dual registration device where both registrations are handled in the same IMS core network, This ENUM enhancement enables SIP requests to be routed to the correct IMS core network (i.e. MMTEL related requests to the IMS core network providing MMTEL services and RCS related requests to the IMS core network providing RCS services).
2. The definition of a UCE Forking and Aggregation (UFA) function to provide the forking and aggregation capability for handling UE Capability Exchange (UCE). This is applicable when the UCE request originates in a single registration device and is required to be sent across the NNI to a device registered in two separate IMS core networks. In this case, before the UCE response is returned to the originating device,

the responses from the two separate IMS core networks should be aggregated. Before the requests are sent to the respective IMS core networks, the originating network may populate the Contact header of the UCE request on behalf of the UE to identify whether the request is related to MMTEL or RCS services, based on IMS registration information. This prevents forking of the UCE when the UCE request is related to MMTEL-only or RCS-only registration. This requirement applies to both OPTIONS and Presence based UCE.

Note: As described in section 2.9 of GSMA PRD NG.102 [10], it is recommended that the UE populate the Contact header of a UCE request with the MMTEL ICSI and/or RCS ICSI/IARI/feature tags.

### C.1 Processing of MMTEL/RCS Requests and Responses

A given service related request (i.e. RCS or MMTEL) must be routed to the correct terminating IMS core network. The following steps must be performed by the network when originating a service request: -

1. Determines via ENUM query whether there are one or two SIP URIs applicable to the target telephone number.
2. Determines by examination of the SIP request, whether it is an MMTEL related request, or an RCS related request (i.e. via the ICSI, IARI and feature tags in SIP INVITE request).
3. Selects the appropriate SIP URI returned in the ENUM response as follows:
  - a. If a single SIP URI is provided in the ENUM result (which denotes a single/converged IMS interconnect), that URI shall be selected
  - b. If two SIP URIs are provided in the ENUM result (denoting separate IMS interconnects for MMTEL and RCS services respectively) and in step 2 the request is found to relate to RCS, the URI provided in the E2U+rsc NAPTR record shall be selected.
  - c. Otherwise (i.e. if two SIP URIs are provided in the ENUM result and the request does not relate to RCS), the URI provided in the E2U+sip NAPTR record shall be selected.

Forwards the message toward the associated network using normal IMS call routing.

Carrier ENUM should be provisioned as follows, based on the services supported by the device to which a given number is assigned.

Services Supported at the NNI	Carrier ENUM Data to Provision
MMTEL and RCS supported on the same IMS core network	One NAPTR specifying the E2U+sip ENUMservice, whose regexp yields a URI resolvable per GSMA PRD IR.67 [23] to an address to which both MMTEL and RCS service requests may be forwarded.
MMTEL and RCS supported on different IMS core networks	Two NAPTRs: One specifying the E2U+sip ENUMservice, whose regexp yields a URI resolvable per GSMA PRD IR.67 [23] to an address to which MMTEL service requests may be forwarded; and

	One specifying the E2U+rcs ENUMservice, whose regexp yields a URI resolvable per GSMA PRD IR.67 [23] to an address to which RCS service requests may be forwarded.
Only MMTEL services supported	One NAPTR specifying the E2U+sip ENUMservice, whose regexp produces a URI resolvable per GSMA PRD IR.67 [23] to an address to which MMTEL service requests may be forwarded.
Only RCS services supported	One NAPTR specifying the E2U+rcs ENUMservice, whose regexp produces a URI resolvable per GSMA PRD IR.67 [23] to an address to which RCS service requests may be forwarded. (NOTE 1)

Note: In this case, if RCS services are provided by an entity not affiliated with the MNO that provides HPMN services to the UE, it may not be possible to identify that MNO from the ENUM response.

There is no special handling required for the SIP response handling (i.e. normal SIP response message routing).

## C.2 Processing of UCE Requests and Responses

UCE requests need special handling when originated by a single registration device and terminated by a dual registration device whose registrations are to two independent IMS core networks. The following steps must be performed by the network when sending the UCE request whether it is OPTIONS based or a Presence based: -

1. If the Contact header in the UCE request contains both the MMTEL ICSI and at least one RCS related ICSI/IARI or feature tag, and if the ENUM query returns two SIP URIs applicable to the target telephone number, then:
  - a. the UFA sends the UCE request towards each of the target URIs. The Contact header of each of the UCE requests shall differ in that they contain either MMTEL services or RCS services.
  - b. The UFA receives the subsequent UCE responses and capabilities via standard SIP message routing mechanisms.
  - c. The UFA must aggregate the service tuples found inside the received UCE responses into one UCE response – see Annex C.3 for details.
  - d. The UFA shall send the combined UCE response using normal IMS call routing towards the originating device.

Otherwise forking is not required. The UFA shall send the UCE request using normal SIP routing as defined in Annex C.1 and there is no special handling required for the subsequent UCE response.

## C.3 Aggregation of UCE Responses at UFA Function

The UE Forking and Aggregation (UFA) function aggregates the service tuples found inside the received UCE responses into one UCE response when: -

The UFA receives responses associated to both forked UCE requests, or

The UFA receives one response with capabilities and SIP Timer F/N is reached (dependent on whether the UCE protocol uses OPTIONS/Presence), the UFA behaves as if a 408/480 response had been received, or

Both SIP specific monitoring timers expire, the UFA behaves as if 408/480 responses had been received.

RCS capabilities are typically cached longer in the UE than MMTel Capabilities. Thus, when an RCS UCE response is unavailable and a MMTEL UCE response is available, it is recommended to send an aggregated UCE response that indicates both RCS UCE and MMTel UCE are unavailable. This enables the UCE query retry mechanism in the UE to take place rather than having the UE interpret a response of just MMTEL capability as an indication that the contact no longer has RCS.

The handling of RCS UCE and MMTel UCE responses are as shown in the table 10 below.

RCS UCE result → ↓ MMTel UCE Result	404/604	480/408	200 OK with capabilities list / no capabilities	Other error
404/604	404/604	480/408	RCS Capabilities / No Capabilities	RCS UCE response
480/408	480/408	480/408	RCS Capabilities / No Capabilities	480/408
200 OK with capabilities list	MMTel Capabilities	480/408 (see Note)	Aggregated MMTel and RCS capabilities	480/408 (see Note)
200 OK with no capabilities	Empty capabilities list	480/408	RCS Capabilities / No Capabilities	480/408
Other error	MMTel UCE response	480/408	RCS Capabilities / No Capabilities	RCS UCE response

**Table 10 – Aggregating UCE Response Handling**

Note: UCE results for RCS are typically cached at the originating UE for much longer than those for MMTel (see section 3.3.1 of GSMA PRD RCC.71 [96]). This would apply also to erroneous results, especially those that provide an incomplete result. Therefore, the table introduces an asymmetry in the error handling that aims to balance shortening the lifetime of incomplete results in the originating UE's cache as much as possible with incorrect indications of service unavailability.

## **Annex D Interworking between UCE Methods (Normative)**

As stated in GSMA PRD IR.90 [5] section 4.1.3, there is a need to provide interworking when the UE Capability Exchange (UCE) mechanism is different between two Operators, i.e., one uses SIP OPTIONS based UCE and the other uses Presence based UCE.

To address this challenge, the following functionality is required to be provided by the network: -

1. The addition of a new function to provide interworking between the UCE based on OPTIONS and UCE based on Presence. This is termed the UPI (UCE Protocol Interworking) function.

### **D.1 UE Protocol Interworking (UPI) Function Triggering and Functionality**

The UPI must be triggered when:

the protocol used in the UCE request to be forwarded is different from the UCE protocol applicable to the NNI (as indicated by the CapabilityDiscoveryMechanism parameter as described in GSMA PRD NG.118 [25]), and

Interworking between different UCE protocols cannot be provided by the NNI (as indicated by the CapabilitiesExchangeInterworking parameter as described in GSMA PRD NG.118 [25]).

Note: If there is a UCE protocol mismatch, but the CapabilitiesExchangeInterworking parameter indicates that the adjacent IMS network can provide the required interworking, then the UPI function may or may not be invoked.

If UCE protocol interworking is required, then the UPI converts the UCE request to send across the NNI as described in section 4.1.3 of GSMA PRD IR.90 [5].

On receipt of a UCE response, the UPI function converts the UCE response to the protocol in which the UCE request was originally received.



## Annex E Example Message Flows for IMS Core Network Interworking and Capability Exchange Interworking (Informative)

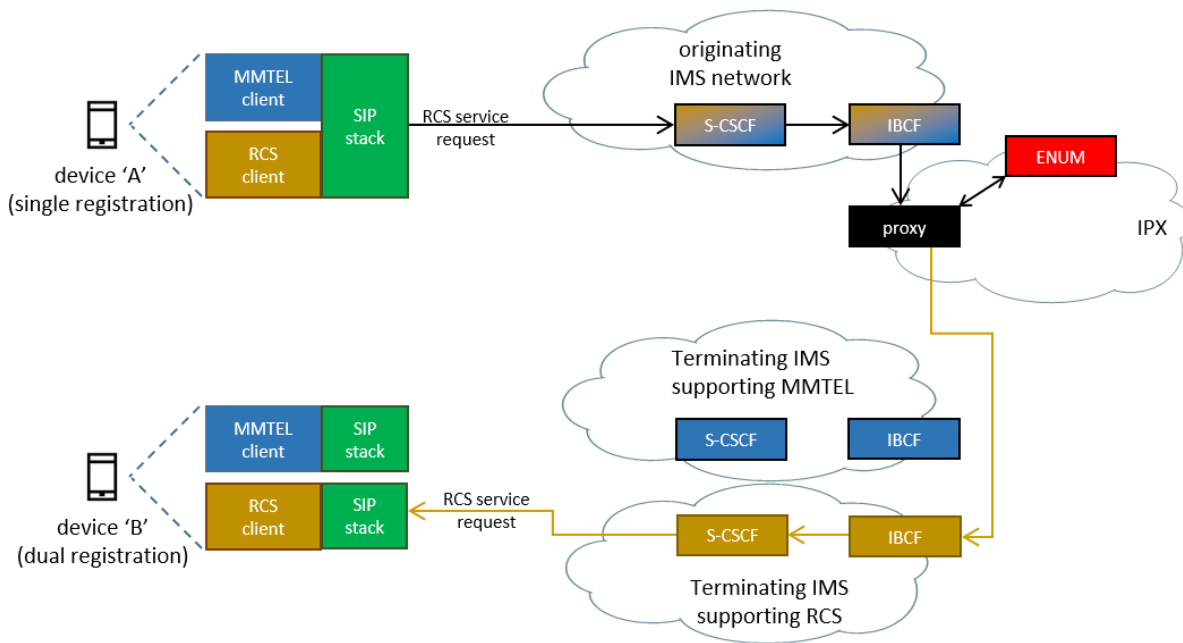
This annex provides an example message flow to illustrate both IMS core network interworking as well as UCE interworking.

The scenarios show interworking between a converged IMS core network and separate IMS core networks. It is also assumed that the UCE protocol is different from one network to the other.

All the message flows show the SIP signalling being conveyed between originating and terminating IMS core networks via an IPX. While the example shows that the ENUM data dip is performed in the IPX that also provides the UFA and UPI functions, any of these functions can also be provided by either network depending on bilateral agreement. It should be noted that these message flows are examples and not proscriptive in any way to alternative implementation decisions.

### E.1 RCS Service Request & Response

Figure 2 below shows the example of a RCS related request being routed from the originating (converged) IMS core network to a terminating (separate IMS core networks).



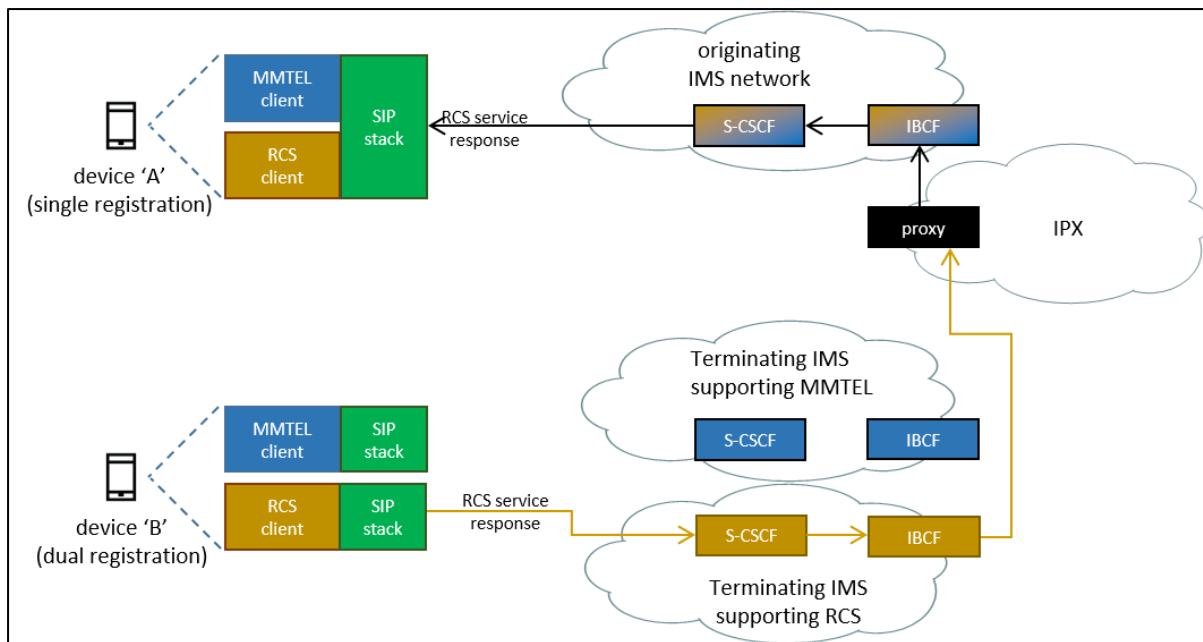
**Figure 2 – Example of interworking RCS service requests between single and dual registration devices**

The message is routed between the originating and terminating IMS networks via IPX. The IPX: -

1. Determines via ENUM query that there are two SIP URIs applicable to the target telephone number.
2. Determines by examination of the request, whether it is an MMTEL or an RCS request (i.e. via ICSI in SIP INVITE request)

3. Selects the appropriate SIP URI returned in the ENUM response, and routes the SIP request towards the associated network using normal IMS call routing. In this case, it is assumed that it is an RCS related request and the message is delivered to the terminating IMS network handling RCS.

Figure 3 illustrates the response message being returned to the originating IMS core network. There is no special handling required in this case (i.e. normal SIP response message routing).



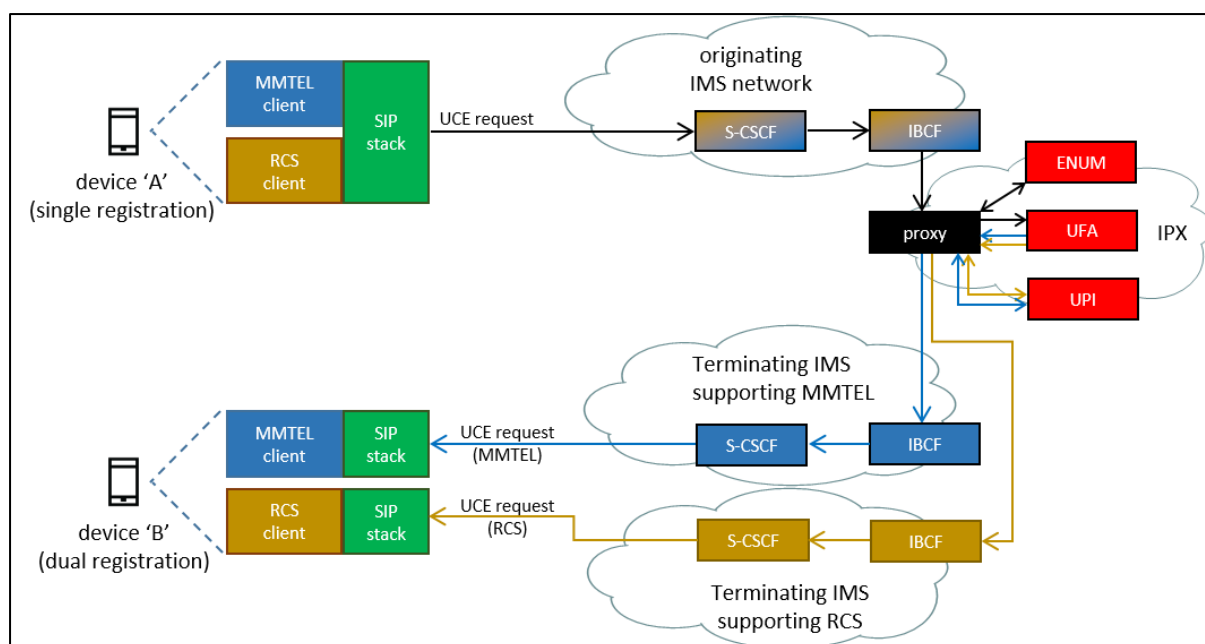
**Figure 3– Example of interworking RCS service responses between single and dual registration devices**

## E.2 Processing of UCE Requests and Responses

Figure 4 shows the UCE request originating from the single IMS core network and must be sent to two different IMS core networks. In addition, there is interworking between the UCE protocols required. Therefore, there is a need for both a UFA and UPI to be invoked.

## E.3 Processing of UCE Requests and Responses

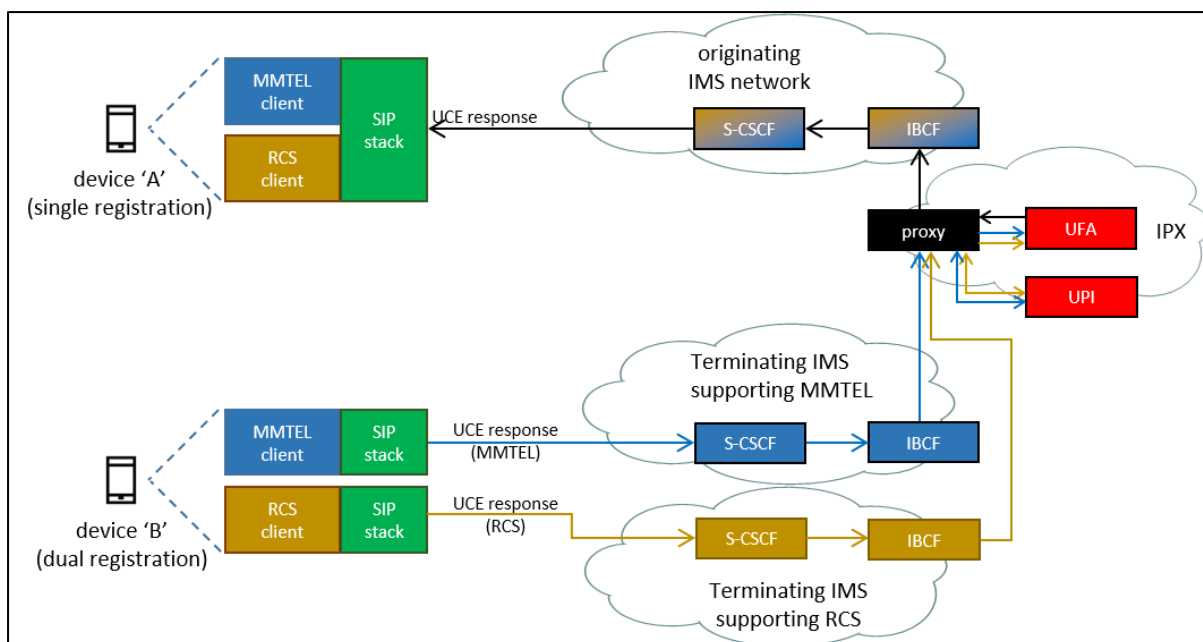
Figure 4 shows the UCE request originating from the single IMS core network and must be sent to two different IMS core networks. In addition, there is interworking between the UCE protocols required. Therefore, there is a need for both a UFA and UPI to be invoked.



**Figure 4- Examples of interworking UCE requests between single and dual registration devices**

The IPX performs the following steps: -

1. Examines the Contact header of the UCE request to determine whether it is related to MMTEL services only, RCS services only or converged (both MMTEL and RCS) services. In this case, it is assumed that the UCE request is interested in converged services.
2. Determines via ENUM query that there are two SIP URIs applicable to the target telephone number. The UCE request must therefore be forked to each of the target URIs. The Contact header of each of the UCE requests differ in that they contain either MMTEL services or RCS services.
3. A check is made to see whether the UPI function is required. In this case, it is assumed that interworking of the UCE protocol is required.
4. The two UCE requests are forwarded to the respective terminating IMS networks.



**Figure 5 Example of interworking UCE responses between single and dual registration devices**

Figure 5 illustrates the processing of response(s) to the UCE requests shown in figure 4. The IPX preforms the following steps -

1. On receipt of a UCE response, the UPI function is invoked to convert the UCE response to the protocol in which the UCE request was originally received and passes on the response to the UFA.
2. The UFA waits to receive both responses and combines the two responses it receives into a single UCE response.
3. The combined UCE response is sent to the originating IMS network using normal IMS call routing towards the originating device.

## Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
.1	7 November 2014	New DRAFT PRD submitted for IREG WG approval	IREG67	Mark McGinley, AT&T
1.0	16 February 2015	PSMC Approval (post PSMC #130)	PSMC	Mark McGinley, AT&T
2.0	22 February 2017	Implemented approved change requests: IR.95 CR 1002 IR.95 CR 1003 IR.95 CR 1004 IR.95 CR 1005 IR.95 CR 1006 IR.95 CR 1007	PSMC	Mark McGinley, AT&T
3.0	29 August 2017	Implemented approved change requests: IR.95 CR 1008 IR.95 CR 1009 IR.95 CR 1010 IR.95 CR 1011	TG (formerly, PSMC)	Mark McGinley, AT&T
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9.0	07 Feb 2024	CR1021- Additions for IMS Data Channel	NG	Javier Sendin, GSMA
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### Other Information

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